DEFENSE INFORMATION SYSTEMS AGENCY
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IN REPLY REFER TO: Joint Interoperability Test Command (JTE) **12 Dec 11**

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Special Interoperability Test Certification of the Avaya Aura™ AS5300 Wide Area Network (WAN) Softswitch (SS), Version 2.0 (with specified patch releases)

References: (a) DoD Directive 4630.05, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004
(b) CJCSI 6212.01E, "Interoperability and Supportability of Information Technology and National Security Systems," 15 December 2008
(c) through (e), see Enclosure 1

1. References (a) and (b) establish the Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification.

2. The Avaya Aura™ AS5300 WAN SS, Version 2.0 (with specified patch releases); hereinafter referred to as the System Under Test (SUT), is certified for joint use within the Defense Information System Network (DISN) as a WAN SS. The Defense Information Systems Agency (DISA) adjudicated all open Test Discrepancy Reports (TDRs) to have a minor operational impact. The fielding of the SUT is limited to IP version 4 (IPv4) across the DISN based on the fielding environment, IP version 6 (IPv6) partial compliance and Plan of Action and Milestones (POA&M) to address the remaining IPv6 discrepancies in their next major release in 2012. DISA retains the authority to remove this product from the Department of Defense (DoD) Unified Capabilities (UC) Approved Products List (APL) as follow-on products are fielded with full IPv6 capability. Any new discrepancy noted in the operational environment will be evaluated for impact on the existing certification. These discrepancies will be adjudicated to the satisfaction of DISA via a vendor POA&M, which will address all new critical TDRs within 120 days of identification. Testing was conducted using WAN SS product requirements derived from the Unified Capabilities Requirements (UCR), Reference (c), and WAN SS test procedures, Reference (d). The SUT was tested with an integrated Local Session Controller (LSC) and is certified for joint use with or without an LSC. No other configurations, features, or functions, except those cited within this memorandum, are certified by JITC. This certification expires upon changes that affect interoperability, but no later than three years from the date the SUT was posted on the Unified Capabilities (UC) Approved Products List (APL) (19 April 2011).

3. This finding is based on interoperability testing conducted by JITC, review of the vendor's Letters of Compliance (LoC), and DISA Chief Information Officer (CIO) approval of the Information Assurance (IA) configuration. Prototype Interoperability testing was conducted by JITC, Fort Huachuca, Arizona, from 29 June through 11 September 2009. APL interoperability testing to UCR 2008 Change 1 requirements was conducted from 13 September through 1 October 2010. Review of the vendor's LoC was completed on 7 October 2010. There were

additional V&V test events conducted on this product to address patch bundle updates. The last V&V test addressed patch bundle 17 and was completed on 4 November 2011. The DISA CIO has reviewed the IA Assessment Report for the SUT, Reference (e), and based on the findings in the report has provided a Authorization to Operate (ATO) on 13 April 2011. The acquiring agency or site will be responsible for the DoD Information Assurance Certification and Accreditation Process (DIACAP) accreditation. Enclosure 2 documents the test results and describes the tested network and system configurations including specified patch releases.

4. The interface, Capability Requirements (CR) and Functional Requirements (FR), and component status of the SUT are listed in Tables 1 and 2. The threshold CR/FR for WAN SSs are established by Section 5.3.2.8.4, 5.3.5, and 5.4 of Reference (c) and were used to evaluate interoperability of the SUT. Enclosure 3 provides a detailed list of WAN SS requirements.

Table 1. SUT Interface Interoperability Status

Interface	Critical	UCR Reference	Threshold CR/FR[1]	Status	Remarks[2]
External Interfaces					
10Base-X	Yes	5.3.2.4.2	1, 2, 3, 4, 5, 6, 7, 9, 10, 11, 12, 13, and 14	Certified	Met threshold CRs/FRs for IEEE 802.3i and 802.3j for the AS-SIP trunk.
100Base-X	Yes	5.3.2.4.2	1, 2, 3, 4, 5, 6, 7, 9, 10, 11, 12, 13, and 14	Certified	Met threshold CRs/FRs for IEEE 802.3u for the AS-SIP trunk.
1000Base-X	Yes	5.3.2.4.2	1, 2, 3, 4, 5, 6, 7, 9, 10, 11, 12, 13, and 14	Certified	Met threshold CRs/FRs for IEEE 802.3z and 802.3ab. Applies to AS-SIP trunk interface.
ISDN T1 PRI ANSI T1.619a	Yes	5.3.2.4.3	2, 3, 6, 7, 9, 11, and 14	Certified	Met threshold CRs/FRs. This interface provides legacy DSN and TELEPORT connectivity.
ISDN T1 PRI NI-2	Yes	5.3.2.4.3	2, 3, 6, 7, 9, 11, and 14	Certified	Met threshold CRs/FRs. This interface provides PSTN connectivity.
E1 PRI ITU-T Q.931	No	5.3.2.12.10	2, 3, 6, 7, 9, 11, and 14	Certified	Met threshold CRs/FRs for this interface. This interface provides PSTN connectivity..
SONET OC-3	No	5.3.2.8.4	2, 3, 6, 7, 9, 11, and 14	Certified	Met threshold CRs/FRs for this interface.
NM					
10Base-X	Yes	5.3.2.4.4 5.3.2.7.2.8	15	Certified	Met threshold CRs/FRs for this interface. Verified via LoC.
100Base-X	Yes	5.3.2.4.4 5.3.2.7.2.8	15	Certified	Met threshold CRs/FRs for this interface. Verified via LoC.

NOTES:
1. The SUT high-level CR and FR ID numbers depicted in the Threshold CRs/FRs column can be cross-referenced in Table 2. These high-level CR/FR requirements refer to a detailed list of requirements provided in Enclosure 3.
2. Detailed information pertaining to open TDRs and associated operational impacts is in Enclosure 2, paragraph 11

Table 1. SUT Interface Interoperability Status (continued)

LEGEND:

10Base-X	10 Mbps Ethernet	JITC	Joint Interoperability Test Command
100Base-X	100 Mbps Ethernet	LoC	Letter of Compliance
1000Base-X	1000 Mbps Ethernet	Mbps	Megabits per second
802.3i	10 Mbps twisted pair media for 10Base-X networks	MLPP	Multi-Level Precedence and Preemption
802.3j	10 Mbps fiber media for 10Base-X networks	NI-2	National ISDN Standard 2
802.3u	100BASE-TX, 100BASE-T4, 100BASE-FX Fast Ethernet at 100 Mbps with auto negotiation	NM	Network Management
		OC-3	Optical Carrier Level 3 (155 Mbps)
ANSI	American National Standards Institute	PRI	Primary Rate Interface
APL	Approved Products List	PSTN	Public Switched Telephone Network
AS-SIP	Assured Services Session Initiation Protocol	Q.931	Signaling Standard for ISDN
CAS	Channel Associated Signaling	Q.955.3	ISDN Signaling Standard for E1 MLPP
CCS7	Common Channel Signaling Number 7	SONET	Synchronous Optical Network
CR	Capability Requirement	SS	Softswitch
DSN	Defense Switched Network	SS7	Signaling System 7
E1	European Basic Multiplex Rate (2.048 Mbps)	SUT	System Under Test
FR	Functional Requirement	T1	Digital Transmission Link Level 1 (1.544 Mbps)
ID	Identification	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
IEEE	Institute of Electrical and Electronics Engineers	TDR	Test Discrepancy Reports
ISDN	Integrated Services Digital Network	UC	Unified Capabilities
ITU-T	International Telecommunication Union – Telecommunication Standardization Sector	UCR	Unified Capabilities Requirements
		WAN	Wide Area Network

Table 2. SUT CR and FR Status

CR/FR ID	Capability/Function	Applicability[1]	UCR Reference	Status	Remarks
1	**Assured Services Product Features and Capabilities**				
	DSCP Packet Marking	Required	5.3.2.2.1.4	Met	None
	Voice Features and Capabilities	Required	5.3.2.2.2.1	Partially Met[2]	None
	Public Safety Features	Required	5.3.2.2.2.2	Met	None
	ASAC Voice	Required	5.3.2.2.2.3.1.2	Met	None
	ASAC Video	Required	5.3.2.2.2.3.2	Met	None
	Signaling Protocols	Required	5.3.2.2.2.3	Met	None
	Signaling Performance	Required	5.3.2.2.2.4	Met	None
2	**Registration, Authentication, and Failover**				
	Registration	Required	5.3.2.3.1	Met	None
	Failover	Required	5.3.2.3.2	Met	None
3	**Product Physical, Quality, and Environmental Factors**				
	Availability	Required	5.3.2.5.2.1	Met	None
	Maximum Downtimes	Required	5.3.2.5.2.2	Met	None
	Loss of Packets	Required	5.3.2.5.4	Met	None
4	**Global Location Server**				
	Global Location Server Requirements	Required	5.3.2.8.2.2	Met	None
5	**LSC Requirements for WAN Softswitch**				
	LSC Requirements	Conditional	5.3.2.7	Partially Met[3]	None

Table 2. SUT CR and FR Status (continued)

CR/FR ID	Capability/Function	Applicability[1]	UCR Reference	Status	Remarks
6	**Call Connection Agent Requirements**				
	CCA IWF Component	Required	5.3.2.9.2.1	Met	None
	CCA MGC Component	Required	5.3.2.9.2.2	Met	None
	SG Component	Conditional	5.3.2.9.2.3	Not Tested[4]	None
	CCA-IWF Support for AS-SIP	Required	5.3.2.9.5.1	Met	None
	CCA-IWF Support for SS7	Conditional	5.3.2.9.5.2	Not Tested[4]	None
	CCA-IWF Support for PRI via MG	Required	5.3.2.9.5.3	Met	None
	CCA-IWF Support for CAS Trunks via MG	Conditional	5.3.2.9.5.4	Not Tested[4]	None
	CCA-IWF Support for VoIP and TDM Protocol Interworking	Required	5.3.2.9.5.6	Met	None
	CCA Preservation of Call Ringing State during Failure Conditions	Required	5.3.2.9.6	Not Met[5]	None
	CCA Interactions with Transport Interface Functions	Required	5.3.2.10.3	Met	None
	CCA Interactions with the EBC	Required	5.3.2.10.4	Met	None
	CCA Support for Admission Control	Required	5.3.2.10.5	Met	None
	CCA Support for UFS	Required	5.3.2.10.6	Met	None
	CCA Support for IA	Required	5.3.2.10.7	Met	None
	CCA Support for AS Voice and Video	Required	5.3.2.10.11	Partially Met[6,7]	None
	CCA Interactions with Service control Functions	Required	5.3.2.10.12	Met	None
	CCA Interworking between AS-SIP and SS7	Conditional	5.3.2.11	Not Tested[4]	None
7	**MG Requirements**				
	Role of MG In SS	Required	5.3.2.12.3.2.1	Met	None
	MG Support for ASAC	Required	5.3.2.12.4.1	Met	None
	MG and IA Functions	Required	5.3.2.12.4.2	Met	None
	MG Interaction with Service Control Function	Required	5.3.2.12.4.3	Met	None
	MG Interactions with IP Transport Interface Functions	Required	5.3.2.12.4.4	Met	None
	MG-EBC interactions	Required	5.3.2.12.4.5	Met	None
	MG IP-Based PSTN Interface Requirements	Conditional	5.3.2.12.4.7	Not Tested[4]	None.
	MG support for User Features and Services	Required	5.3.2.12.4.9	Met	None
	MG Interface to TDM	Required	5.3.2.12.5	Met[4]	None
	MG Interface to TDM Allied and Coalition	Conditional	5.3.2.12.6	Not Tested[4]	None.
	MG Interface to TDM PSTN in U.S	Required	5.3.2.12.7	Met	None
	MG Interfaces to TDM PSTN OCONUS	Required	5.3.2.12.8	Met	None
	MG Support for CCS7	Conditional	5.3.2.12.9	Not Tested[4]	None
	MG Support for ISDN PRI Trunks	Required	5.3.2.12.10	Met	None
	MG Support for CAS Trunks	Conditional	5.3.2.12.11	Not Tested[4]	None
	MG Echo Cancellation	Required	5.3.2.12.13	Met	None
	MG Clock Timing	Required	5.3.2.12.14	Met	None
	MGC-MG CCA Functions	Required	5.3.2.12.15	Met	None
	MG V.150.1	Required	5.3.2.12.16	Not Met[8]	None
	MG Preservation of Call Ringing during Failure	Required	5.3.2.12.17	Not Tested[5]	None

Table 2. SUT CR and FR Status (continued)

CR/FR ID	Capability/ Function	Applicability[1]	UCR Reference	Status	Remarks
8	**SG Requirements**				
	SG and CCS7 Network Interactions	Conditional	5.3.2.13.5.1	Not Tested[4]	None
	SG Interactions with CCA	Conditional	5.3.2.13.5.2	Not Tested[4]	None
	SG Interworking Functions	Conditional	5.3.2.13.5.3	Not Tested[4]	None
9	**WWNDP Requirements**				
	WWNDP	Required	5.3.2.16	Met	None
	DSN WWNDP	Required	5.3.2.16.1	Met	None
10	**Commercial Cost Avoidance**				
	Commercial Cost Avoidance	Required	5.3.2.23	Met[9]	None
11	**Precedence Call Diversion**				
	Precedence call Diversion	Conditional	5.3.2.25	Met	None
12	**AS-SIP Requirements**				
	AS-SIP General Requirements	Required	5.3.4	Partially Met[7]	None
	SIP Session Keep-Alive Timer	Required	5.3.4.8	Met	None
	Session Description Protocol	Required	5.3.4.9	Met	None
	Precedence and Preemption	Required	5.3.4.10	Met	None
	Video Telephony – General Rules	Required	5.3.4.12	Partially Met[7]	None
	Calling Services	Required	5.3.4.13	Met	None
	SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances	Required	5.3.4.14	Met	None
	Relevant Timers for the Terminating Gateway and the Originating Gateway	Required	5.3.4.15	Met	None
	SIP Requirements for Interworking AS-SIP Signaling Appliance	Required	5.3.4.16	Met	None
	Keep-Alive Timer Requirements for Interworking AS-SIP Signaling Appliances	Required	5.3.4.17	Met	None
	Precedence and Preemption Extensions for Interworking AS-SIP Signaling Appliances	Required	5.3.4.18	Met	None
	Supplementary Services	Required	5.3.4.19	Met	None
13	**IPv6 Requirements**				
	Product Requirements	Required	5.3.5.4	Partially Met[10]	None
14	**Information Assurance**				
	Information Assurance Requirements	Required	5.4	Met[11]	None
15	**Network Management**				
	General Management Requirements	Required	5.3.2.17.2	Partially Met[12]	None
	VVoIP NMS Interface Requirements	Required	5.3.2.4.4	Partially Met[12]	None
	Requirement for FCAPS Management	Required	5.3.2.17.3	Partially Met[12,13]	None
	NM requirements of Appliance Functions	Required	5.3.2.18	Partially Met[12]	None
	Accounting Management	Required	5.3.2.19	Partially Met[12]	None

Table 2. SUT CR and FR Status (continued)

NOTES:

1. The annotation of 'required' refers to a high-level requirement category. The applicability of each sub-requirement is provided in Enclosure 3.
2. The SUT had outstanding open TDRs at the completion of testing, which were adjudicated by DISA to have a minor operational impact. The vendor has submitted a POA&M to address the open TDRs. Reference (f), Enclosure 2, Paragraph 11, provides additional details.
3. The LSC is an optional integrated component of the SUT and; therefore, the SUT is certified for joint use with or without the LSC. The SUT was certified with noted minor operational discrepancies. The LSC Special Interoperability Certification letter and test summary report is posted on the UC APL under TN# 0911801. The SUT partially met PEI requirements (no video). The AEI and Operator Console requirements were not tested; this requirement represents a new UCR requirement (Jan 2010) at the time of the APL interoperability testing and therefore compliance is not mandatory at that time, based on allowance of an 18-month development cycle for new requirements.
4. This capability or interface is a conditional requirement for a WAN SS. The SUT met all the interfaces requirements for a T1 ISDN PRI (ANSI T1.619a and ANSI T1 607 NI2) and E1 ISDN PRI (ETSI PSTN interface only).
5. This requirement represents a new UCR requirement (Jan 2010) at the time of the APL interoperability testing and therefore compliance is not mandatory at that time, based on allowance of an 18-month development cycle for new requirements.
6. The SUT PEI hardphone met the UCR requirements for voice only. The PEI softphone met both voice and video requirements with one exception: The softphone can assign any DSCP value from 0-63 to meida and signaling but cannot assign a unique DSCP value for each precedence level per the UCR when running on Windows Vista or Windows 7. The softphone assigns the same DSCP value for all precedence levels. This discrepancy was adjudicated by DISA on Autust 2011 with a minor operationl impact.
7. The vendor did not support AEI video or voice capability. This was adjudicated by DISA to have a minor operational impact since there were no certified AEI video end instruments on the UC APL and furthermore, AEIs are a new UCR 2008, Change 1 requirement and therefore compliance is not mandatory at the time of APL interoperability testing, based on allowance of an 18-month development cycle for new requirements.
8. The vendor did not demonstrate V.150.1 support. This requirement represents a new UCR requirement (Jan 2010) at the time of the APL interoperability testing and therefore compliance is not mandatory at that time, based on allowance of an 18-month development cycle for new requirements.
9. The SUT met this requirement with a Lightweight Directory Access Protocol server which is covered under a separate Interoperability Certification listed separately on the UC APL.
10. The DISA adjudicated all open TDRs to have a minor operational impact. The fielding of the SUT is limited to IPv4 across the DISN based on the fielding environment, IPv6 partial compliance and POA&M addressing critical IPv6 discrepancies in their next major release in 2012. DISA retains the authority to remove this product from the Department of Defense (DoD) Unified Capabilities (UC) Approved Products List (APL) as follow-on products are fielded with full IPv6 capability. The SUT was tested and met IPv6 interoperability requirements with its optional LSC intra-enclave only with the following discrepancies which were adjudicated by DISA as having an minor operational impact:
    a. POA&M. The SUT does not meet RFC 4007 for IPv6 Scoped Address Architecture.
    b. The SUT does not support IPv6 (Signaling or Media) with the MP112 and MP124 analog IADs.
    c. The SUT SESM Core supports IPv4 only for signaling inter-enclave (WAN).
    d. The SUT Audio Codes MG3K supports IPv4 only for signaling and both IPv4 and IPv6 dual stack for media intra and interenclave.
11. Information Assurance was tested by a DISA-led Information Assurance test team and published in a separate report, Reference (e).
12. The vendor submitted a NM LoC with noted discrepancies. The following open TDRs were adjudicated by DISA to have a minor operational impact with a vendor submitted POA&M:
    a. The SUT does not fully support SNMP and MIBs IAW IETF Standards 58 and 62.
    b. The SUT is not fully compliant with NM call detail records formats.
    c. SUT does not support management requirements for ASAC.
13. The SUT does not support destination code controls. The SUT does not have the capability of setting the percentage of calls to be blocked to the designated destination(s). This was adjudicated by DISA to have a minor operational impact.

Table 2. SUT CR and FR Status (continued)

LEGEND:

AEI	Assured Services End Instrument
APL	Approved Products List
ASAC	Assured Services Admission Control
AS	Assured Services
ASD/NII	Assistant Secretary of Defense for Networks and Information Integration
AS-SIP	Assured Services Session Initiation Protocol
CAS	Channel Associated Signaling
CCA	Call Connection Agent
CCS7	Common Channel Signaling Number 7
CR	Capability Requirement
CM	Configuration Management
DISA	Defense Information Systems Agency
DISN	Defense Information System Network
DoD	Department of Defense
DSCP	Differentiated Services Code Point
DSN	Defense Switched Network
E1	European Basic Multiplex Rate (2.048 Mbps)
EMS	Element Management System
FR	Functional Requirement
IA	Information Assurance
IAW	In accordance with
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPSec	Internet Protocol Security
IPv6	Internet Protocol version 6
ISDN	Integrated Services Digital Network
IWF	Interworking Function
LDAP	Lightweight Directory Access Protocol
LoC	Letter of Compliance
LSC	Local Session Controller
Mbps	Megabits per second
MG	Media Gateway
MGC	Media Gateway Controller
MIB	Management Information Base
NM	Network Management
NMS	Network Management System
OCONUS	Outside the Continental United States
PEI	Proprietary End Instrument
POA&M	Plan of Action and Milestones
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
RFC	Request for Comment
SESM	Subscriber Edge Services Manager
SG	Signaling Gateway
SIP	Session Initiation Protocol
SNMP	Simple Network Management Protocol
SNMPv2	Simple Network Management Protocol version 2
SNMPv3	Simple Network Management Protocol version 3
SS	Softswitch
SS7	Signaling System 7
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
TDR	Test Discrepancy Report
TDM	Time Division Multiplexing
UC	Unified Capabilities
UCR	Unified Capabilities Requirements
VoIP	Voice over Internet Protocol
VVoIP	Voice and Video over Internet Protocol
WAN	Wide Area Network
WWNDP	World Wide Numbering and Dialing Plan

5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi. All associated data is available on the Defense Information Systems Agency Unified Capability Coordination Office (UCCO) website located at http://www.disa.mil/ucco/.

6. The JITC point of contact is Captain Stéphane Arsenault, JITC, commercial (520) 538-5269 or DSN 312-879-5269; e-mail address is Stephane.Arsenault@disa.mil. The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The UCCO tracking number for the SUT is 1031901.

FOR THE COMMANDER:

J. T. Schutte

3 Enclosures a/s

for BRADLEY A. CLARK
Chief
Battlespace Communications Portfolio

Distribution (electronic mail):
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Joint Interoperability Test Command, Liaison, TE3/JT1
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Department of the Army, Office of the Secretary of the Army, DA-OSA CIO/G-6 ASA (ALT), SAIS-IOQ
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DOT&E, Net-Centric Systems and Naval Warfare
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Defense Information Systems Agency, TEMC
Office of Assistant Secretary of Defense (NII)/DOD CIO
U.S. Joint Forces Command, Net-Centric Integration, Communication, and Capabilities Division, J68
Defense Information Systems Agency, GS23

ADDITIONAL REFERENCES

(c) Office of the Assistant Secretary of Defense, "Department of Defense Unified Capabilities Requirements 2008, Change 1," 22 January 2010
(d) Joint Interoperability Test Command, "Unified Capabilities Test Plan (UCTP)"
(e) Joint Interoperability Test Command, "Information Assurance (IA) Assessment of Avaya Aura™ AS5300 Version 2.0 WAN SS (TN 1031901)," 1 April 2011

CERTIFICATION TESTING SUMMARY

1. SYSTEM TITLE. Special Interoperability Test Certification of the Avaya Aura™ AS5300 Wide Area Network (WAN) Softswitch (SS), Version 2.0 (with specified patch releases)

2. SPONSOR. Defense Information Systems Agency (DISA), Attention: Louis Schmuckler GS15, Capability Center, IP Division Chief, GS23, PO Box 4502, Arlington VA, 22204-4502, Phone 703-882-0274, e-mail: louis.schmuckler@disa.mil.

3. SYSTEM POC. Avaya Government Solutions, Attention: William Stehling, Address: 12730 Fair Lakes Circle, Fairfax, VA, 22033-4901, Phone: 703-539-3109, e-mail: William.Stehling@avayagov.com.

4. TESTER. Joint Interoperability Test Command (JITC), Fort Huachuca, Arizona.

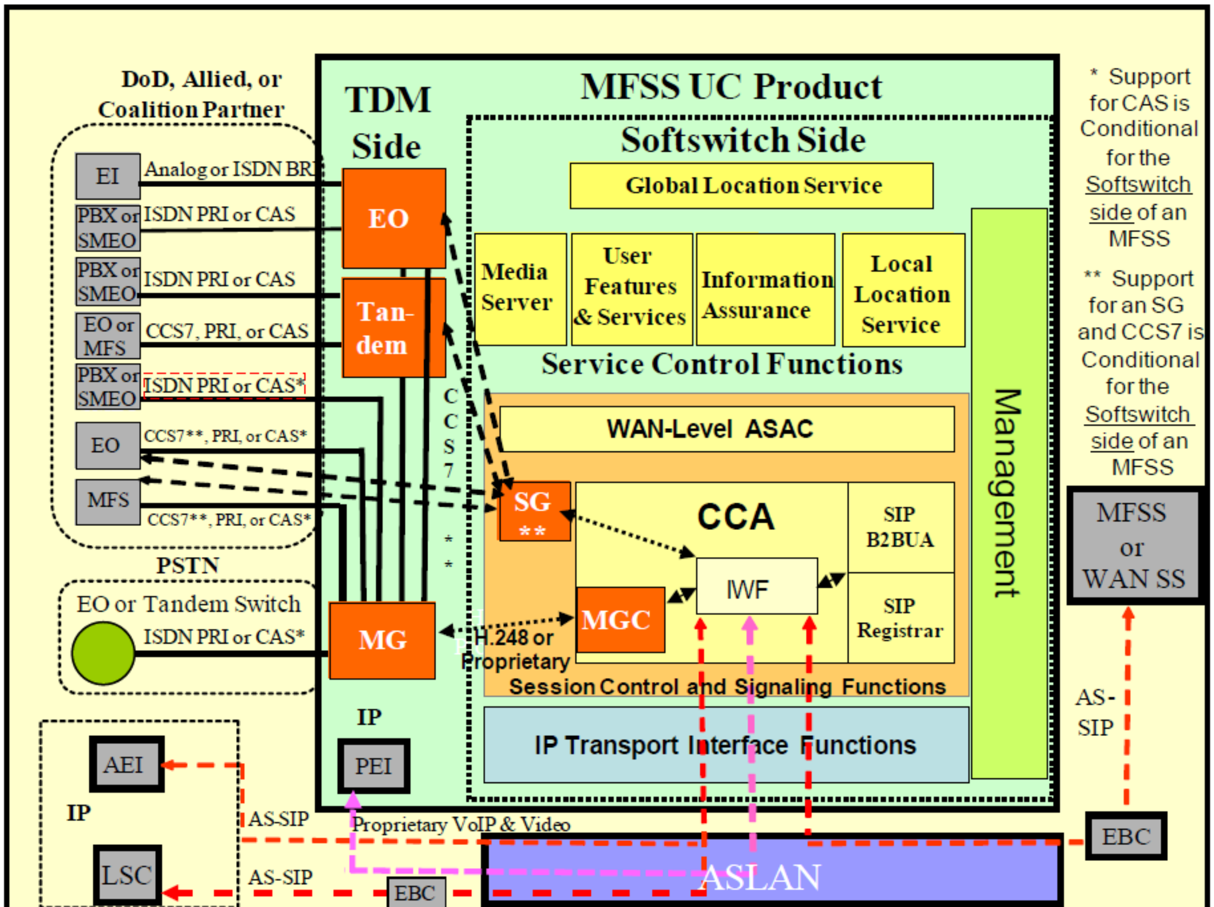
5. SYSTEM DESCRIPTION. The Avaya Aura AS5300 Wide Area Network (WAN) Softswitch (SS), hereafter referred to as the System Under Test (SUT) is a Session Initiation Protocol (SIP)-based session manager designed to increase productivity and collaboration by allowing users to collaborate using the appropriate tool (instant messaging, chat, voice, video, file transfer, web collaboration, etc.) all in an integrated solution. The AS5300 supports the SIP, Assured Services SIP (AS-SIP), Secure Real-time Transport Protocol (SRTP), Session Description Protocol (SDP) Security Descriptions for Media Streams and Transport Layer Security (TLS). The AS5300 encrypts both the voice conversation and the signaling setup to create a secure Voice-over-IP (VoIP) environment.

The SUT interfaces to legacy Multi-Function Switches via the Mediant 3000 Trunking Gateway Optical Carrier Level 3 (155Mbps). JITC only tested the Mediant Gateway 3000 (MG3K) Trunking Gateway during this event with OC3 V1.5 T1 Integrated Services Digital Network (ISDN) Primary Rate Interfaces (PRI) interfaces. The SUT was also tested with direct E1 copper interfaces. JITC analysis determined that the T1 ISDN PRI copper interface of the MG3K only differs at layer 1 and is similar to the OC3 for interoperability purposes and is also certified for joint use within the DISN. The following interfaces are certified with the SUT ISDN PRIs supporting ANSI T1.619a ISDN PRI NI2 (Assured Services), ANSI T1.607 NI2 (Public Switched Telephone Network (PSTN) only) interfaces, and European Telecommunications Standard Institute (ETSI) E1 ISDN PRI (PSTN only).

The SUT servers use Ethernet interfaces that are connected to the service provider's Data Communication Network (DCN), and is configured using Provisioning Manager (PROV) System Manager Graphical user Interface. All management traffic to and from network elements and to and from clients and higher-level management systems travel over these interfaces. The SUT consists of the following component:

AudioCodes Mediant™ 3000 Media Gateway. The AudioCodes Mediant™ 3000 is a SIP Network Gateway that supports Inter-switch SIP to Time Division Multiplixing (TDM) gateway call types. The SIP Network Gateway allows carriers to evolve their session/signaling control network with a hierarchy, separating the call agent server who services the subscribers, from the SIP call agent that hosts Intra-domain (or intra-network) interfaces, or Inter-domain (or inter-network) interfaces that may or may not use a Session Border Controller.

6. OPERATIONAL ARCHITECTURE. Figure 2-1 depicts the WAN SS functional model and Figure 2-2 the notional operational architecture that the SUT may be used in.

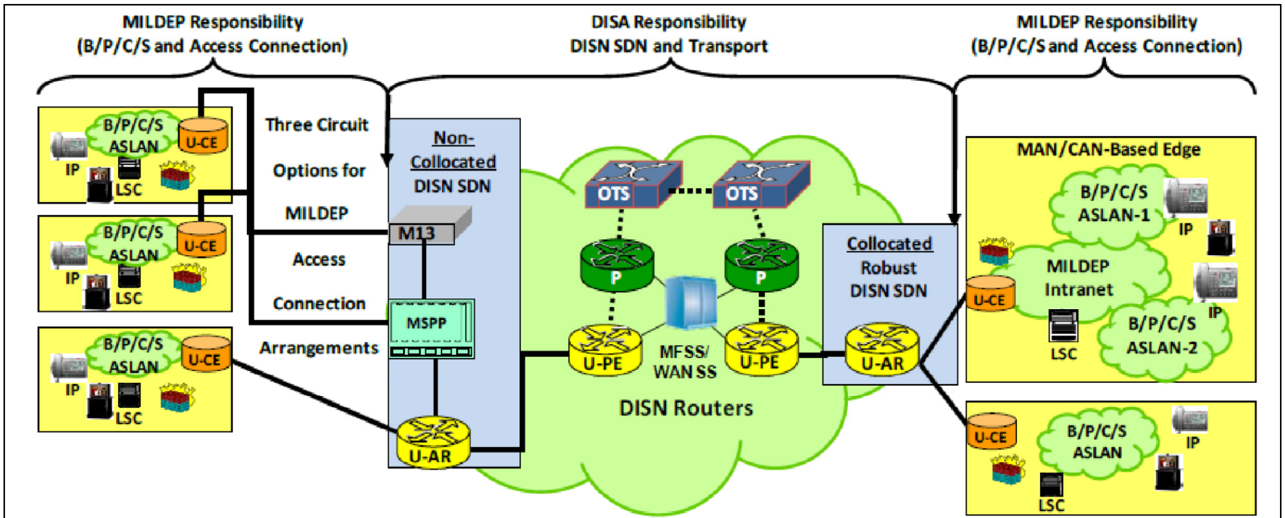


LEGEND:

AEI	AS-SIP End Instrument	LSC	Local Session Controller
ASAC	Assured Service Admission Control	MFS	Multifunction Switch
ASLAN	Assured Services Local Area Network	MFSS	Multifunction Softswitch
AS-SIP	Assured Services Session Initiation Protocol	MG	Media Gateway
BRI	Basic Rate Interface	MGC	Media Gateway Controller
B2BUA	Back-to-Back User Agent	PBX	Private Branch Exchange
CAS	Channel Associated Signaling	PEI	Proprietary End Instrument
CCA	Call Connection Agent	PRI	Primary Rate Interface
CCS7	Common Channel Signaling No. 7	PSTN	Public Switched Telephone Network
DoD	Department of Defense	SCS	Session Control and Signaling
EBC	Edge Border Controller	SG	Signaling Gateway
EI	End Instrument	SIP	Session Initiation Protocol
EO	End Office	SMEO	Small End Office
IA	Information Assurance	SS	Softswitch
IP	Internet Protocol	SUT	System Under Test
ISDN	Integrated Services Digital Network	TDM	Time Division Multiplexer
IWF	Interworking Function	VoIP	Voice over Internet Protocol
		WAN	Wide Area Network

Note: The PEI (using Proprietary VoIP and Video) is part of the MFSS UC Product; the AEI (using AS-SIP VoIP and Video) is not.

Figure 2-1. WAN SS Functional Reference Model



LEGEND:

ASLAN	Assured Services Local Area Network	MILDEP	Military Department
B/P/C/S	Base, Post, Camp, Station	MSPP	Multi-Service Provisioning Platform
CAN	Campus Area Network	OTS	Optical Transport System
DISA	Defense Information Systems Agency	P	Provider
DISN	Defense Information System Network	SDN	Service Delivery Node
IP	Internet Protocol	SS	Softswitch
LSC	Local Session Controller	U-AR	Unclassified Aggregation Router
M13	Multiplexer	U-CE	Unclassified Customer Edge Router
MAN	Metropolitan Area Network	U-PE	Unclassified Provider Edge Router
MFSS	Multifunction Softswitch	WAN	Wide Area Network

Figure 2-2. SUT Notional Operational Architecture

7. INTEROPERABILITY REQUIREMENTS. The interface, Capability Requirements (CR) and Functional Requirements (FR), Information Assurance (IA), and other requirements for WAN SSs are established by Section 5.3.2.8.4, 5.3.5, and 5.4 of Reference (c).

7.1 Interfaces. The SUT uses external interfaces to connect to the Global Information Grid network and other Unified Capabilities products. Table 2-1 shows the physical interfaces supported by the SUT and the associated standards.

Table 2-1. SUT Interface Requirements

Interface	Critical	UCR Reference	Threshold CR/FR (see Note)	Remarks
External Interfaces				
10Base-X	Yes	5.3.2.4.2	1, 2, 3, 4, 5, 6, 7, 9, 10, 11, 12, 13, and 14	None
100Base-X	Yes	5.3.2.4.2	1, 2, 3, 4, 5, 6, 7, 9, 10, 11, 12, 13, and 14	None
1000Base-X	Yes	5.3.2.4.2	1, 2, 3, 4, 5, 6, 7, 9, 10, 11, 12, 13, and 14	None
ISDN T1 PRI ANSI T1.619a	Yes	5.3.2.4.3	2, 3, 6, 7, 9, 11, and 14	None
ISDN T1 PRI NI-2	Yes	5.3.2.4.3	2, 3, 6, 7, 9, 11, and 14	None
E1 PRI ITU-T Q.931	No	5.3.2.12.10	2, 3, 6, 7, 9, 11, and 14	None
SONET OC-3	No	5.3.2.8.4	2, 3, 6, 7, 9, 11, and 14	None
NM				
10Base-X	Yes	5.3.2.4.4 5.3.2.7.2.8	15	None
100Base-X	Yes	5.3.2.4.4 5.3.2.7.2.8	15	None

NOTE: The SUT high-level CR and FR ID numbers depicted in the Threshold CRs/FRs column can be cross-referenced in Table 2. These high-level CR/FR requirements refer to a detailed list of requirements provided in Enclosure 3.

LEGEND:

10Base-X	10 Mbps Ethernet
100Base-X	100 Mbps Ethernet
1000Base-X	1000 Mbps Ethernet
ANSI	American National Standards Institute
CR	Capability Requirement
E1	European Basic Multiplex Rate (2.048 Mbps)
FR	Functional Requirement
ID	Identification
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union – Telecommunication Standardization Sector
NI-2	National ISDN Standard 2
NM	Network Management
OC-3	Optical Carrier Level 3 (155 Mbps)
PRI	Primary Rate Interface
Q.931	Signaling Standard for ISDN
SONET	Synchronous Optical Network
T1	Digital Transmission Link Level 1 (1.544 Mbps)
T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
UCR	Unified Capabilities Requirements

7.2 CR and FR. The WAN SSs have required and conditional features and capabilities that are established by Section 5.3.2.8.4, 5.3.5, and 5.4 of Reference (c). The SUT does not need to provide non-critical (conditional) requirements. If they are provided, they must function according to the specified requirements. The SUTs features and capabilities and its aggregated requirements are listed in Table 2-2. Detailed CR/FR requirements are provided in Table 3-1 of Enclosure 3.

Table 2-2. SUT CRs and FRs

CR/FR ID	Capability/Function	Applicability[1]	UCR Reference	Remarks
1	**Assured Services Product Features and Capabilities**			
	DSCP Packet Marking	Required	5.3.2.2.1.4	None
	Voice Features and Capabilities	Required	5.3.2.2.2.1	None
	Public Safety Features	Required	5.3.2.2.2.2	None
	ASAC Voice	Required	5.3.2.2.2.3.1.2	None
	ASAC Video	Required	5.3.2.2.2.3.2	None
	Signaling Protocols	Required	5.3.2.2.2.3	None
	Signaling Performance	Required	5.3.2.2.2.4	None
2	**Registration, Authentication, and Failover**			
	Registration	Required	5.3.2.3.1	None
	Failover	Required	5.3.2.3.2	None
3	**Product Physical, Quality, and Environmental Factors**			
	Availability	Required	5.3.2.5.2.1	None
	Maximum Downtimes	Required	5.3.2.5.2.2	None
	Loss of Packets	Required	5.3.2.5.4	None
4	**Global Location Server**			
	Global Location Server Requirements	Required	5.3.2.8.2.2	None
5	**LSC Requirements for WAN Softswitch**			
	LSC Requirements	Conditional	5.3.2.7	None
6	**CCA Requirements**			
	CCA IWF Component	Required	5.3.2.9.2.1	None
	CCA MGC Component	Required	5.3.2.9.2.2	None
	SG Component	Conditional	5.3.2.9.2.3	None
	CCA-IWF Support for AS-SIP	Required	5.3.2.9.5.1	None
	CCA-IWF Support for SS7[2]	Conditional	5.3.2.9.5.2	None
	CCA-IWF Support for PRI via MG	Required	5.3.2.9.5.3	None
	CCA-IWF Support for CAS Trunks via MG[2]	Conditional	5.3.2.9.5.4	None
	CCA-IWF Support for VoIP and TDM Protocol Interworking	Required	5.3.2.9.5.6	None
	CCA Preservation of Call Ringing State during Failure Conditions	Required[5]	5.3.2.9.6	None
	CCA Interactions with Transport Interface Functions	Required	5.3.2.10.3	None
	CCA Interactions with the EBC	Required	5.3.2.10.4	None
	CCA Support for Admission Control	Required	5.3.2.10.5	None
	CCA Support for UFS	Required	5.3.2.10.6	None
	CCA Support for IA	Required	5.3.2.10.7	None
	CCA Support for AS Voice and Video[3]	Required	5.3.2.10.11	None
	CCA Interactions with Service control Functions	Required	5.3.2.10.12	None
	CCA Interworking between AS-SIP and SS7	Conditional	5.3.2.11	None

Table 2-2. SUT CRs and FRs (continued)

CR/FR ID	Capability/Function	Applicability[1]	UCR Reference	Remarks
7	**MG Requirements**			
	Role of MG In SS	Required	5.3.2.12.3.2.1	None
	MG Support for ASAC	Required	5.3.2.12.4.1	None
	MG and IA Functions	Required	5.3.2.12.4.2	None
	MG Interaction with Service Control Function	Required	5.3.2.12.4.3	None
	MG Interactions with IP Transport Interface Functions	Required	5.3.2.12.4.4	None
	MG-EBC interactions	Required	5.3.2.12.4.5	None
	MG IP-Based PSTN Interface Requirements	Conditional	5.3.2.12.4.7	None
	MG support for User Features and Services	Required	5.3.2.12.4.9	None
	MG Interface to TDM	Required	5.3.2.12.5	None
	MG Interface to TDM Allied and Coalition	Conditional	5.3.2.12.6	None
	MG Interface to TDM PSTN in U.S.	Required	5.3.2.12.7	None
	MG Interfaces to TDM PSTN OCONUS	Required	5.3.2.12.8	None
	MG Support for CCS7[3]	Conditional	5.3.2.12.9	None
	MG Support for ISDN PRI Trunks	Required	5.3.2.12.10	None
	MG Support for CAS Trunks	Conditional	5.3.2.12.11	None
	MG Echo Cancellation	Required	5.3.2.12.13	None
	MG Clock Timing	Required	5.3.2.12.14	None
	MGC-MG CCA Functions	Required	5.3.2.12.15	None
	MG V.150.1[3]	Required	5.3.2.12.16	None
	MG Preservation of Call Ringing during Failure[3]	Required	5.3.2.12.17	None
8	**SG Requirements**			
	SG and CCS7 Network Interactions	Conditional	5.3.2.13.5.1	None
	SG Interactions with CCA	Conditional	5.3.2.13.5.2	None
	SG Interworking Functions	Conditional	5.3.2.13.5.3	None
9	**WWNDP Requirements**			
	WWNDP	Required	5.3.2.16	None
	DSN WWNDP	Required	5.3.2.16.1	None
10	**Commercial Cost Avoidance**			
	Commercial Cost Avoidance	Required	5.3.2.23	None
11	**Precedence Call Diversion**			
	Precedence call Diversion	Conditional	5.3.2.25	None
12	**AS-SIP Requirements**			
	AS-SIP General Requirements	Required	5.3.4	None
	SIP Session Keep-Alive Timer	Required	5.3.4.8	None
	Session Description Protocol	Required	5.3.4.9	None
	Precedence and Preemption	Required	5.3.4.10	None
	Video Telephony – General Rules	Required	5.3.4.12	None
	Calling Services	Required	5.3.4.13	None
	SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances	Required	5.3.4.14	None
	Relevant Timers for the Terminating Gateway and the Originating Gateway	Required	5.3.4.15	None
	SIP Requirements for Interworking AS-SIP Signaling Appliance	Required	5.3.4.16	None

Table 2-2. SUT CR and FR (continued)

CR/FR ID	Capability/ Function	Applicability[1]	UCR Reference	Remarks
12	**AS-SIP Requirements (continued)**			
	Keep-Alive Timer Requirements for Interworking AS-SIP Signaling Appliances	Required	5.3.4.17	None
	Precedence and Preemption Extensions for Interworking AS-SIP Signaling Appliances	Required	5.3.4.18	None
	Supplementary Services	Required	5.3.4.19	None
13	**IPv6 Requirements**			
	Product Requirements	Required	5.3.5.4	None
14	**Information Assurance**			
	Information Assurance Requirements	Required	5.4	None
15	**Network Management**			
	General Management Requirements	Required	5.3.2.17.2	None
	VVoIP NMS Interface Requirements	Required	5.3.2.4.4	None
	Requirement for FCAPS Management	Required	5.3.2.17.3	None
	NM requirements of Appliance Functions	Required	5.3.2.18	None
	Accounting Management	Required	5.3.2.19	None

NOTES:
1. The annotation of 'required' refers to a high-level requirement category. The applicability of each sub-requirement is provided in Enclosure 3.
2. This capability or interface is a conditional requirement for a WAN SS.
3. This requirement represents a new UCR requirement (Jan 2010) at the time of the APL interoperability testing and therefore compliance is not mandatory at that time, based on allowance of an 18-month development cycle for new requirements.

LEGEND:

APL	Approved Products List
ASAC	Assured Services Admission Control
AS	Assured Services
AS-SIP	Assured Services Session Initiation Protocol
CAS	Channel Associated Signaling
CCA	Call Connection Agent
CCS7	Common Channel Signaling Number 7
CR	Capability Requirement
CM	Configuration Management
DISA	Defense Information Systems Agency
DSCP	Differentiated Services Code Point
DSN	Defense Switched Network
EBC	Edge Boundary Controller
EMS	Element Management System
FCAPS	Fault, Configuration, Accounting, Performance, and Security
FR	Functional Requirement
IA	Information Assurance
IPv6	Internet Protocol version 6
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
IWF	Interworking Function
LoC	Letter of Compliance
LSC	Local Session Controller
Mbps	Megabits per second
MG	Media Gateway
MGC	Media Gateway Controller
NM	Network Management
NMS	Network Management System
OCONUS	Outside the Continental United States
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
SG	Signaling Gateway
SIP	Session Initiation Protocol
SS	Softswitch
SS7	Signaling System 7
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
TDM	Time Division Multiplexing
UCR	Unified Capabilities Requirements
VoIP	Voice over Internet Protocol
VVoIP	Voice and Video over Internet Protocol
WAN	Wide Area Network
WWNDP	World Wide Numbering and Dialing Plan

7.3 IA. Table 2-3 details the IA requirements applicable to a WAN SS.

Table 2-3. SS IA Requirements

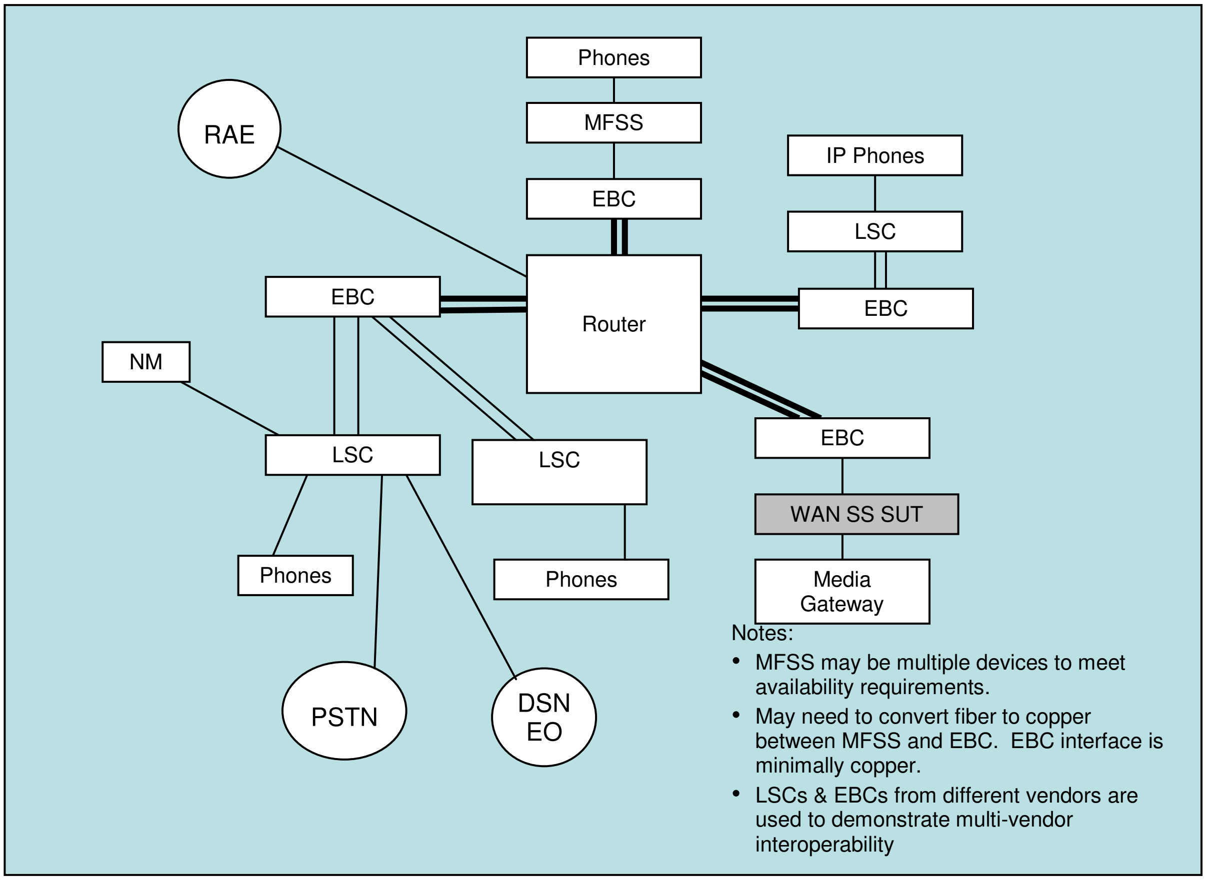
Requirement	Applicability (See note)	UCR Reference	Criteria
General Requirements	Required	5.4.6.2	Detailed requirements and associated criteria for Softswitches are listed in the IATP, Reference (e).
Authentication	Required	5.4.6.2.1	
Integrity	Required	5.4.6.2.2	
Confidentiality	Required	5.4.6.2.3	
Non-Repudiation	Required	5.4.6.2.4	
Availability	Required	5.4.6.2.5	

NOTE:
The annotation of 'Required' refers to the high-level requirement category. The applicability of each sub-requirement is provided in Reference (e).

LEGEND:
IA Information Assurance
IATP IA Test Plan
UCR Unified capabilities Requirements

7.4 Other. None

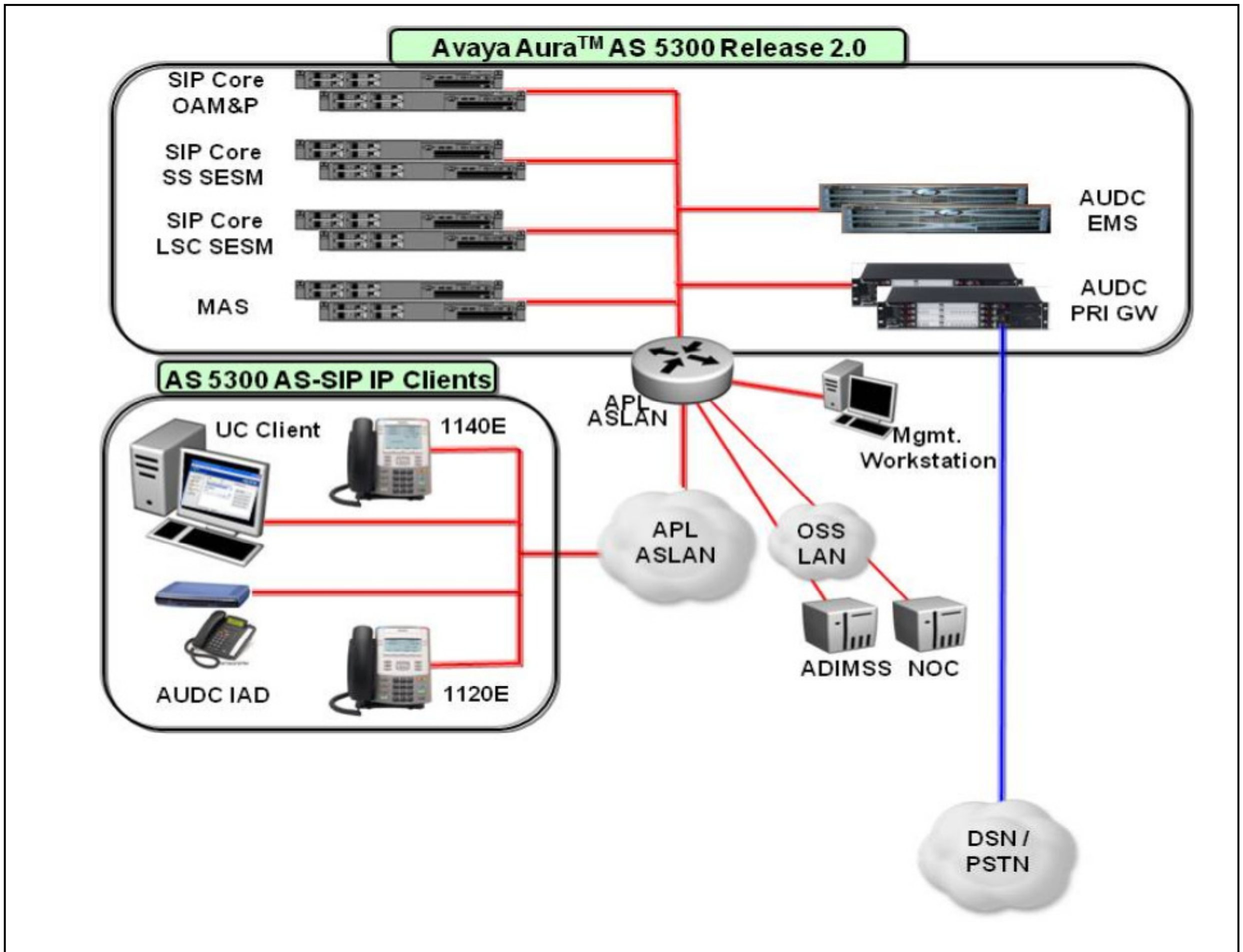
8. TEST NETWORK DESCRIPTION. The SUT was tested at JITC in a manner and configuration similar to that of a notional operational environment. Testing of the system's required functions and features was conducted using the test configurations depicted in Figures 2-3 and 2-4. Figure 2-3 depicts the minimum test architecture for testing a WAN SS. Figure 2-4 depicts the SUT's test configuration.



LEGEND:

C	Conditional	NM	Network Management
DSN	Defense Switched Network	PSTN	Public Switch Telephone Network
EBC	Edge Boundary Controller	R	Required
EO	End Office	RAE	Required Ancillary Equipment
IP	Internet Protocol	SUT	System Under Test
LSC	Local Session Controller	WAN SS	Wide Area Network Softswitch
MFSS	MultiFunction Softswitch		

Figure 2-3. WAN SS Minimum Test Architecture



LEGEND:

ADIMSS	Advanced DSN Integrated Management Support System
APL	Approved Products List
AS	Assured Services
AS-LAN	Assured Services Local Area Network
AUDC	Audio Codes
DSN	Defense Switched Network
EMS	Element Management System
GW	Gateway
IAD	Integrated Access Device
LAN	Local Area Network
MAS	Media Application Server
Mgmt	Management
NOC	Network Operations Control
OAM&P	Operations Administration Maintenance & Provisioning
OSS	Operational Support System
PRI	Primary Rate Interface
PSTN	Public Switch Telephone Network
SESM	Session Management
SIP	Session Initiation Protocol
SS	Softswitch
SWEX	Switch Expert

Figure 2-4. SUT Test Configuration

9. SYSTEM CONFIGURATIONS. Table 2-4 provides the system configurations and hardware and software components tested with the SUT. The SUT was tested in an operationally realistic environment to determine its interoperability capability with associated network devices and network traffic.

Table 2-4. Tested System Configurations

<table>
<tr><th colspan="2">System Name</th><th>Software Release</th></tr>
<tr><td colspan="2">Avaya AS5300 LSC</td><td>AS5300 Release 2.0
Load: MCP_13.0.0.0_2010-05-05-2108
Patch: MCP_13.0.0.14_2011-10-06-0305, patch bundle 17</td></tr>
<tr><td colspan="2">Cisco CUCM LSC</td><td>8.0(2)</td></tr>
<tr><td colspan="2">Avaya Aura S8800 LSC</td><td>Communication Manager 6.0.1 (00.1.510.1 Service Pack 19211)</td></tr>
<tr><td colspan="2">Nokia Siemens Networks HiQ8000 LSC</td><td>13.90.02.10 Patch Set (PS) 14, Patch (P) 102</td></tr>
<tr><th colspan="3">SUT Components</th></tr>
<tr><th colspan="3">Switch Expert</th></tr>
<tr><th>Part Number</th><th>Part Description</th><th>Firmware/Software</th></tr>
<tr><td>NTVW02DP</td><td>AS5300 R2.0 Switch Expert MFSS Software CD</td><td>Version 7.0, Build 360</td></tr>
<tr><td>NTVW02DQ</td><td>AS5300 R2.0 Switch Expert Small Package Software CD</td><td>Version 7.0, Build 360</td></tr>
<tr><td>NTVW02DN</td><td>AS5300 Release 2.0 Switch Expert MFSS Package</td><td>Version 7.0, Build 360</td></tr>
<tr><td>NTVW02DO</td><td>AS5300 Release 2.0 Switch Expert Small Package</td><td>Version 7.0, Build 360</td></tr>
<tr><th colspan="3">SIP Core OAM&P</th></tr>
<tr><th>Part Number</th><th>Part Description</th><th>Firmware/Software</th></tr>
<tr><td>NTVW02AD</td><td>AS5300 Release 2.0 SIP Core New System Software Package</td><td rowspan="3">Load: MCP_13.0.0.5_2010-10-07-1208, patch:
MCP_13.0.0.14_2011-10-06-0305 patch bundle 17
Oracle version: 10.2.0.4.0, patch level 23</td></tr>
<tr><td>NTVW02AF</td><td>AS5300 R1.0 to R2.0 Upgrade w/SRS PrePaid</td></tr>
<tr><td>NTVW02AG</td><td>AS5300 R1.0 to R2.0 Upgrade and Expansions</td></tr>
<tr><td>NTVW02DE</td><td>AS5300 Release 2.0 OS Software Kit - CDROM and DVD</td><td>Version: 13.0.19
Patch to: 13.0.29</td></tr>
<tr><td>NTVW02DD</td><td>AS5300 Release 2.0 Core Apps Software Kit CDROM and DVD</td><td>Load: MCP_13.0.0.5_2010-10-07-1208, patch:
MCP_13.0.0.14_2011-10-06-0305 patch bundle 17
Oracle version: 10.2.0.4.0, patch level 23</td></tr>
<tr><th colspan="3">SIP Core SS Session Manager</th></tr>
<tr><th>Part Number</th><th>Part Description</th><th>Firmware/Software</th></tr>
<tr><td>NTVW02AD</td><td>AS5300 Release 2.0 SIP Core New System Software Package</td><td rowspan="3">Load: MCP_13.0.0.5_2010-10-07-1208, patch:
MCP_13.0.0.9_2011-05-20-1232 patch bundle 12
Oracle version: 10.2.0.4.0, patch level 21</td></tr>
<tr><td>NTVW02AF</td><td>AS5300 R1.0 to R2.0 Upgrade w/SRS PrePaid</td></tr>
<tr><td>NTVW02AG</td><td>AS5300 R1.0 to R2.0 Upgrade and Expansions</td></tr>
<tr><td>NTVW02DE</td><td>AS5300 Release 2.0 OS Software Kit - CDROM and DVD</td><td>Version: 13.0.19
Patch to: 13.0.29</td></tr>
<tr><td>NTVW02DD</td><td>AS5300 Release 2.0 Core Apps Software Kit CDROM and DVD</td><td>Load: MCP_13.0.0.5_2010-10-07-1208, patch:
MCP_13.0.0.14_2011-10-06-0305 patch bundle 17
Oracle version: 10.2.0.4.0, patch level 23</td></tr>
<tr><th colspan="3">Media Application Server</th></tr>
<tr><th>Part Number</th><th>Part Description</th><th>Firmware/Software</th></tr>
<tr><td>NTVW02AE</td><td>AS5300 Release 2.0 MAS New System Software Package</td><td>Version: 6.6.0.88</td></tr>
<tr><td>NTVW02AF</td><td>AS5300 R1.0 to R2.0 Upgrade w/SRS PrePaid</td><td>Version: 6.6.0.88</td></tr>
<tr><td>NTVW02AG</td><td>AS5300 R1.0 to R2.0 Upgrade and Expansions</td><td>Version: 6.6.0.88</td></tr>
<tr><td>NTVW02DF</td><td>AS5300 Rls 2.0 MAS Base and Applications</td><td>Version: 6.6.0.88</td></tr>
<tr><td>NTVW02DE</td><td>AS5300 Release 2.0 OS Software Kit - CDROM and DVD</td><td>Version: 13.0.19
Patch to: 13.0.29</td></tr>
</table>

Table 2-4. Tested System Configurations (continued)

Audio Codes M3K Gateway		
Part Number	**Part Description**	**Firmware/Software**
NTVW02CF	AS5300 Release 2.0 Audiocodes M3K PRI GW 8 Span Bundle DC	Version: 5.80A.045.000
NTVW02CG	AS5300 Release 2.0 Audiocodes M3K PRI GW 8 Span Bundle AC	Version: 5.80A.045.000
NTVW02CH	AS5300 Release 2.0 Audiocodes M3K PRI GW 12 Span Bundle DC	Version: 5.80A.045.000
NTVW02CI	AS5300 Release 2.0 Audiocodes M3K PRI GW 12 Span Bundle AC	Version: 5.80A.045.000
NTVW02CJ	AS5300 Release 2.0 Audiocodes M3K PRI GW 16 Span Bundle DC	Version: 5.80A.045.000
NTVW02CK	AS5300 Release 2.0 Audiocodes M3K PRI GW 16 Span Bundle AC	Version: 5.80A.045.000
NTVW02CL	AS5300 Release 2.0 Audiocodes M3K PRI GW 42 Span Bundle DC	Version: 5.80A.045.000
NTVW02CM	AS5300 Release 2.0 Audiocodes M3K PRI GW 42 Span Bundle AC	Version: 5.80A.045.000
NTVW02DG	AS5300 Release 2.0 Audiocodes M3K PRI GW Software CD	Version: 5.80A.045.000
NTVW02CN	AS5300 Release 2.0 Audiocodes M3K PRI GW OC3 Bundle DC	Version: 5.80A.045.000
NTVW02CO	AS5300 Release 2.0 Audiocodes M3K PRI GW OC3 Bundle AC	Version: 5.80A.045.000
NTVW02DH	AS5300 Release 2.0 Audiocodes M3K/6310 PRI GW Software CD	Version: 5.80A.045.000
Audiocodes Element Management System		
Part Number	**Part Description**	**Firmware/Software**
NTVW02AY	AS5300 Audiocodes EMS Server Package AC	Version: 5.8.83
NTVW02AZ	AS5300 Audiocodes EMS Server Package DC	
NTVW02DK	AS5300 Release 2.0 Audiocodes EMS Application SW on Sun Netra T2000 Server with Oracle 9.2 and Solaris 10 and JAVA ES installed	
Client Workstations		
Device Name		**Firmware/Software**
CPE AS5300 UC Client (previously MMPC Client)		Version: 7.2.3075_20110728[1]
Management Workstation		Customer Provided STIG'd PC
Avaya Aura AS5300 LSC Software release 2.0 (with specified patch bundle)[2]		

Notes:
1. The UC client is certified with Windows XP, Vista, and Win7 operation systems
2. Refer to the Avaya Aura AS5300 with software version 2.0 LSC Interoperability Certification Letter and Test Summary Report and associated Desktop Review extensions listed on the UC APL, Reference (g). The Avay Aura AS5300 LSC is an optional component of the SUT and the SUT can be purchased with or without the Avaya Aura AS5300 LSC.

LEGEND:

APL	Approved Product List
CD	Compact Disc
CDROM	Compact Disc Read Only Memory
CPE	Customer Provided Equipment
DVD	Digital Video Disc
EMS	Element Management System
GW	Gateway
LSC	Local Session Controller
M3K	Mediant 3000
MAS	Media Application Server
MCP	Media Communications Processor
MMPC	Modular Messaging PC Client
MFSS	Multi-Function Softswitch
OAM&P	Operations, Administration, Maintenance, and Provisioning
OS	Operating System
PC	Personal Computer
PRI	Primary Rate Interface
R	Release
SIP	Session Initiation Protocol
SRS	Seamless Roaming Phase
SS	Softswitch
STIG	Security Technical Implementation Guides
SUT	System Under Test
UC	Unified Capabilities

10. TESTING LIMITATIONS. The JITC test team noted the following testing limitations including the impact they may have on interpretation of the results and conclusions. Any untested requirements are also included in the testing limitations.

a. Call Loading. Due to limitations in test equipment JITC could not create a large volume of line and trunk calls to simulate operational traffic loads. This issue will be resolved in the near future to allow for simulated call loads during interoperability certification testing. The use of operational data as the SUT is fielded will validate the SUTs ability to support its proposed number of subscribers (up to 250,000).

b. AS-SIP End Instruments (AEI). JITC did not test the SUT with generic AEIs because none were available at the time of test. A vendor has yet to submit a product as an AEI for certification. This requirement represents a new UCR requirement (Jan 2010) at the time of the APL interoperability testing and therefore compliance is not mandatory at that time, based on allowance of an 18-month development cycle for new requirements.

c. Proprietary End Instruments (PEI). JITC did not test hardphone PEIs for video requirements. The only devices tested for video were PEI Softphones. The video phones were tested intra-enclave and inter-enclave between the SUTs Local Session Controller (LSC) and other Avaya Aura AS5300 LSCs. There are currently no other LSC vendors that offer a video end instrument and; therefore, multi-vendor interoperability has not been demonstrated.

d. Internet Protocol version 6 (IPv6). JITC tested the SUT to verify its IPv6 capabilities intra-enclave (with integrated LSC). Inter-enclave End to End IPv6 testing is scheluled for fourth quarter of 2011. The vendor submitted a letter of compliance (LoC) stating limitations. Paragraph 11 provides detailed information about IPv6 results.

e. Network Management (NM). JITC did not test the SUTs NM capabilities to meet UCR requirements. The vendor did submit an NM LoC that was reviewed by JITC. The JITC's evaluation of the SUT's NM capabilities is provided in paragraph 11.

f. Secure Data and Secure Voice Calls. Since the standard for modem over IP is based on International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) V.150.1, secure calls could not be tested inter-enclave (between LSCs/WAN SSs and Mulltifunction Soft Switches (MFSS) via DISN). This requirement represents a new UCR requirement (Jan 2010) at the time of the APL interoperability testing and therefore compliance is not mandatory at that time, based on allowance of an 18-month development cycle for new requirements.

11. INTEROPERABILITY EVALUATION RESULTS. The SUT meets the critical interoperability requirements for a WAN SS in accordance with the UCR and is certified for joint use with other UC Products listed on the APL. Additional discussion regarding specific testing results is located in subsequent paragraphs.

11.1 Interfaces. The SUT met the external interface requirements for 10/100/1000Base-X (AS-SIP line and trunk) and 10/100Base-X for NM. It also met T1 ISDN PRI for both ANSI T1.619a (Assured Services) and ANSI T1.607 (PSTN only) with National ISDN-2 (NI-2), and ETSI E1 ISDN PRI Q.931 (PSTN only) interfaces. The interface status of the SUT is provided in Table 2-5.

Table 2-5. SUT Interface Interoperability Status

Interface	Critical	UCR Reference	Threshold CR/FR[1]	Status	Remarks[2]
External Interfaces					
10Base-X	Yes	5.3.2.4.2	1, 2, 3, 4, 5, 6, 7, 9, 10, 11, 12, 13, and 14	Certified	Met threshold CRs/FRs for IEEE 802.3i and 802.3j for the AS-SIP trunk.
100Base-X	Yes	5.3.2.4.2	1, 2, 3, 4, 5, 6, 7, 9, 10, 11, 12, 13, and 14	Certified	Met threshold CRs/FRs for IEEE 802.3u for the AS-SIP trunk.
1000Base-X	Yes	5.3.2.4.2	1, 2, 3, 4, 5, 6, 7, 9, 10, 11, 12, 13, and 14	Certified	Met threshold CRs/FRs for IEEE 802.3z and 802.3ab. Applies to AS-SIP trunk interface.
ISDN T1 PRI ANSI T1.619a	Yes	5.3.2.4.3	2, 3, 6, 7, 9, 11, and 14	Certified	Met threshold CRs/FRs. This interface provides legacy DSN and TELEPORT connectivity.
ISDN T1 PRI NI-2	Yes	5.3.2.4.3	2, 3, 6, 7, 9, 11, and 14	Certified	Met threshold CRs/FRs. This interface provides PSTN connectivity.
E1 PRI ITU-T Q.931	No	5.3.2.12.10	2, 3, 6, 7, 9, 11, and 14	Certified	Met threshold CRs/FRs for this interface. This interface provides PSTN connectivity..
SONET OC-3	No	5.3.2.8.4	2, 3, 6, 7, 9, 11, and 14	Certified	Met threshold CRs/FRs for this interface.
NM					
10Base-X	Yes	5.3.2.4.4 5.3.2.7.2.8	15	Certified	Met threshold CRs/FRs for this interface. Verified via LoC.
100Base-X	Yes	5.3.2.4.4 5.3.2.7.2.8	15	Certified	Met threshold CRs/FRs for this interface. Verified via LoC.

NOTES:
1. The SUT high-level CR and FR ID numbers depicted in the Threshold CRs/FRs column can be cross-referenced in Table 2. These high-level CR/FR requirements refer to a detailed list of requirements provided in Enclosure 3.
2. Detailed information pertaining to open TDRs and associated operational impacts is in Enclosure 2, paragraph 11

LEGEND:

10Base-X	10 Mbps Ethernet
100Base-X	100 Mbps Ethernet
1000Base-X	1000 Mbps Ethernet
802.3i	10 Mbps twisted pair media for 10Base-X networks
802.3j	10 Mbps fiber media for 10Base-X networks
802.3u	100BASE-TX, 100BASE-T4, 100BASE-FX Fast Ethernet at 100 Mbps with auto negotiation
ANSI	American National Standards Institute
APL	Approved Products List
AS-SIP	Assured Services Session Initiation Protocol
CAS	Channel Associated Signaling
CCS7	Common Channel Signaling Number 7
CR	Capability Requirement
DSN	Defense Switched Network
E1	European Basic Multiplex Rate (2.048 Mbps)
FR	Functional Requirement
ID	Identification
IEEE	Institute of Electrical and Electronics Engineers
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union – Telecommunication Standardization Sector
JITC	Joint Interoperability Test Command
LoC	Letter of Compliance
Mbps	Megabits per second
MLPP	Multi-Level Precedence and Preemption
NI-2	National ISDN Standard 2
NM	Network Management
OC-3	Optical Carrier Level 3 (155 Mbps)
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
Q.931	Signaling Standard for ISDN
Q.955.3	ISDN Signaling Standard for E1 MLPP
SONET	Synchronous Optical Network
SS	Softswitch
SS7	Signaling System 7
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
TDRs	Test Discrepancy Reports
UC	Unified Capabilities
UCR	Unified Capabilities Requirements
WAN	Wide Area Network

11.2 CR and FR. The SUT CR and FR status is depicted in Table 2-6. Detailed CR/FR requirements are provided in Enclosure 3, Table 3-1. A summary of the SUT's ability to meet UCR requirements are provided in the sub-paragraphs below. All requirements and associated references were derived from UCR 2008 Change 1. Discrepancies discussed below were adjudicated to be minor based on vendor submission and compliance to a Plan of Actions and Milestones (POA&M).

Table 2-6. SUT CRs and FRs Status

CR/FR ID	Capability/Function	Applicability[1]	UCR Reference	Status	Remarks
1	**Assured Services Product Features and Capabilities**				
	DSCP Packet Marking	Required	5.3.2.2.1.4	Met	None
	Voice Features and Capabilities	Required	5.3.2.2.2.1	Partially Met[2]	None
	Public Safety Features	Required	5.3.2.2.2.2	Met	None
	ASAC Voice	Required	5.3.2.2.2.3.1.2	Met	None
	ASAC Video	Required	5.3.2.2.2.3.2	Met	None
	Signaling Protocols	Required	5.3.2.2.2.3	Met	None
	Signaling Performance	Required	5.3.2.2.2.4	Met	None
2	**Registration, Authentication, and Failover**				
	Registration	Required	5.3.2.3.1	Met	None
	Failover	Required	5.3.2.3.2	Met	None
3	**Product Physical, Quality, and Environmental Factors**				
	Availability	Required	5.3.2.5.2.1	Met	None
	Maximum Downtimes	Required	5.3.2.5.2.2	Met	None
	Loss of Packets	Required	5.3.2.5.4	Met	None
4	**Global Location Server**				
	Global Location Server Requirements	Required	5.3.2.8.2.2	Met	None
5	**LSC Requirements for WAN Softswitch**				
	LSC Requirements	Conditional	5.3.2.7	Partially Met[3]	None
6	**Call Connection Agent Requirements**				
	CCA IWF Component	Required	5.3.2.9.2.1	Met	None
	CCA MGC Component	Required	5.3.2.9.2.2	Met	None
	SG Component	Conditional	5.3.2.9.2.3	Not Tested[4]	None
	CCA-IWF Support for AS-SIP	Required	5.3.2.9.5.1	Met	None
	CCA-IWF Support for SS7	Conditional	5.3.2.9.5.2	Not Tested[4]	None
	CCA-IWF Support for PRI via MG	Required	5.3.2.9.5.3	Met	None
	CCA-IWF Support for CAS Trunks via MG	Conditional	5.3.2.9.5.4	Not Tested[4]	None
	CCA-IWF Support for VoIP and TDM Protocol Interworking	Required	5.3.2.9.5.6	Met	None
	CCA Preservation of Call Ringing State during Failure Conditions	Required	5.3.2.9.6	Not Met[5]	None
	CCA Interactions with Transport Interface Functions	Required	5.3.2.10.3	Met	None

Table 2-6. SUT CR and FR Status (continued)

CR/FR ID	Capability/ Function	Applicability[1]	UCR Reference	Status	Remarks
6	**Call Connection Agent Requirements (continued)**				
	CCA Interactions with the EBC	Required	5.3.2.10.4	Met	None
	CCA Support for Admission Control	Required	5.3.2.10.5	Met	None
	CCA Support for UFS	Required	5.3.2.10.6	Met	None
	CCA Support for IA	Required	5.3.2.10.7	Met	None
	CCA Support for AS Voice and Video	Required	5.3.2.10.11	Partially Met[6,7]	None
	CCA Interactions with Service control Functions	Required	5.3.2.10.12	Met	None
	CCA Interworking between AS-SIP and SS7	Conditional	5.3.2.11	Not Tested[4]	None
7	**MG Requirements**				
	Role of MG In SS	Required	5.3.2.12.3.2.1	Met	None
	MG Support for ASAC	Required	5.3.2.12.4.1	Met	None
	MG and IA Functions	Required	5.3.2.12.4.2	Met	None
	MG Interaction with Service Control Function	Required	5.3.2.12.4.3	Met	None
	MG Interactions with IP Transport Interface Functions	Required	5.3.2.12.4.4	Met	None
	MG-EBC interactions	Required	5.3.2.12.4.5	Met	None
	MG IP-Based PSTN Interface Requirements	Conditional	5.3.2.12.4.7	Not Tested[4]	None.
	MG support for User Features and Services	Required	5.3.2.12.4.9	Met	None
	MG Interface to TDM	Required	5.3.2.12.5	Met[4]	None
	MG Interface to TDM Allied and Coalition	Conditional	5.3.2.12.6	Not Tested[4]	None.
	MG Interface to TDM PSTN in U.S.	Required	5.3.2.12.7	Met[4]	None
	MG Interfaces to TDM PSTN OCONUS	Required	5.3.2.12.8	Met	None
	MG Support for CCS7	Conditional	5.3.2.12.9	Not Tested[4]	None
	MG Support for ISDN PRI Trunks	Required	5.3.2.12.10	Met	None
	MG Support for CAS Trunks	Conditional	5.3.2.12.11	Not Tested[4]	None
	MG Echo Cancellation	Required	5.3.2.12.13	Met	None
	MG Clock Timing	Required	5.3.2.12.14	Met	None
	MGC-MG CCA Functions	Required	5.3.2.12.15	Met	None
	MG V.150.1	Required	5.3.2.12.16	Not Tested[8]	None
	MG Preservation of Call Ringing during Failure	Required	5.3.2.12.17	Not Tested[5]	None
8	**SG Requirements**				
	SG and CCS7 Network Interactions	Conditional	5.3.2.13.5.1	Not Tested[4]	None
	SG Interactions with CCA	Conditional	5.3.2.13.5.2	Not Tested[4]	None
	SG Interworking Functions	Conditional	5.3.2.13.5.3	Not Tested[4]	None
9	**WWNDP Requirements**				
	WWNDP	Required	5.3.2.16	Met	None
	DSN WWNDP	Required	5.3.2.16.1	Met	None
10	**Commercial Cost Avoidance**				
	Commercial Cost Avoidance	Required	5.3.2.23	Met[9]	None
11	**Precedence Call Diversion**				
	Precedence call Diversion	Conditional	5.3.2.25	Met	Required if SUT includes LSC

Table 2-6. SUT CR and FR Status (continued)

CR/FR ID	Capability/ Function	Applicability[1]	UCR Reference	Status	Remarks
12	**AS-SIP Requirements**				
	AS-SIP General Requirements	Required	5.3.4	Partially Met[7]	None
	SIP Session Keep-Alive Timer	Required	5.3.4.8	Met	None
	Session Description Protocol	Required	5.3.4.9	Met	None
	Precedence and Preemption	Required	5.3.4.10	Met	None
	Video Telephony – General Rules	Required	5.3.4.12	Partially Met[7]	None
	Calling Services	Required	5.3.4.13	Met	None
	SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances	Required	5.3.4.14	Partially Met	None
	Relevant Timers for the Terminating Gateway and the Originating Gateway	Required	5.3.4.15	Met	None
	SIP Requirements for Interworking AS-SIP Signaling Appliance	Required	5.3.4.16	Met	None
	Keep-Alive Timer Requirements for Interworking AS-SIP Signaling Appliances	Required	5.3.4.17	Met	None
	Precedence and Preemption Extensions for Interworking AS-SIP Signaling Appliances	Required	5.3.4.18	Met	None
	Supplementary Services	Required	5.3.4.19	Met	None
13	**IPv6 Requirements**				
	Product Requirements	Required	5.3.5.4	Partially Met[10]	None
14	**Information Assurance**				
	Information Assurance Requirements	Required	5.4	Met[11]	None
15	**Network Management**				
	General Management Requirements	Required	5.3.2.17.2	Partially Met[12]	None
	VVoIP NMS Interface Requirements	Required	5.3.2.4.4	Partially Met[12]	None
	Requirement for FCAPS Management	Required	5.3.2.17.3	Partially Met[12,13]	None
	NM requirements of Appliance Functions	Required	5.3.2.18	Partially Met[12]	None
	Accounting Management	Required	5.3.2.19	Partially Met[12]	None

Table 2-6. SUT CRs and FRs Status (continued)

NOTES:

1. The annotation of 'required' refers to a high-level requirement category. The applicability of each sub-requirement is provided in Enclosure 3.
2. The SUT had outstanding open TDRs at the completion of testing, which were adjudicated by DISA to have a minor operational impact. The vendor has submitted a POA&M to address the open TDRs. Reference (f), Enclosure 2, Paragraph 11, provides additional details.
3. The LSC is an optional integrated component of the SUT and therefore the SUT is certified for joint use with or without the LSC. The SUT was certified with noted minor operational discrepancies. The LSC Special Interoperability Certification letter and test summary report is posted on the UC APL under TN# 0911801. The SUT partially met PEI requirements (no video). The AEI and Operator Console requirements were not tested; this requirement represents a new UCR requirement (Jan 2010) at the time of the APL interoperability testing and therefore compliance is not mandatory at that time, based on allowance of an 18-month development cycle for new requirements..
4. This capability or interface is a conditional requirement for a WAN SS. The SUT met all the interfaces requirements for a T1 ISDN PRI (ANSI T1.619a, and ANSI T1 607 NI2) and E1 ISDN PRI (ETSI PSTN interface only).
5. This requirement represents a new UCR requirement (Jan 2010) at the time of the APL interoperability testing and therefore compliance is not mandatory at that time, based on allowance of an 18-month development cycle for new requirements.6. The SUT PEI hardphone met the UCR requirements for voice only. The PEI softphone met both voice and video requirements with one exception: The softphone can assign any DSCP value from 0-63 to meida and signaling but cannot assign a unique DSCP value for each precedence level per the UCR when running on Windows Vista or Windows 7. The softphone assigns the same DSCP value for all precedence levels. This discrepancy was adjudicated by DISA on Autust 2011 with a minor operationl impact.
7. The vendor did not support AEI video or voice capability. This was adjudicated by DISA to have a minor operational impact since there were no certified AEI video end instruments on the UC APL and furthermore, AEIs are a new UCR 2008, Change 1 requirement and therefore compliance is not mandatory at the time of APL interoperability testing, based on allowance of an 18-month development cycle for new requirements.
8. The vendor did not demonstrate V.150.1 support. This requirement represents a new UCR requirement (Jan 2010) at the time of the APL interoperability testing and therefore compliance is not mandatory at that time, based on allowance of an 18-month development cycle for new requirements.
9. The SUT met this requirement with a Lightweight Directory Access Protocol server which is covered under a separate Interoperability Certification listed separately on the UC APL.
10. The Defense Information Systems Agency (DISA) adjudicated all open Test Discrepancy Reports (TDRs) to have a minor operational impact. The fielding of the SUT is limited to IPv4 across the DISN based on the fielding environment, IPv6 partial compliance and POA&M addressing critical IPv6 discrepancies in their next major release in 2012. DISA retains the authority to remove this product from the Department of Defense (DoD) Unified Capabilities (UC) Approved Products List (APL) as follow-on products are fielded with full IPv6 capability. The SUT was tested and met IPv6 interoperability requirements with its optional LSC intra-enclave only with the following discrepancies which were adjudicated by DISA as having an minor operational impact:
    a. The SUT does not meet RFC 4007 for IPv6 Scoped Address Architecture.
    b. The SUT does not support IPv6 (Signaling or Media) with the MP112 and MP124 analog IADs.
    c. The SUT SESM Core supports IPv4 only for signaling inter-enclave (WAN).
    d. The SUT Audio Codes MG3K supports IPv4 only for signaling and both IPv4 and IPv6 dual stack for media intra and interenclave.
11. Information Assurance was tested by a DISA-led Information Assurance test team and published in a separate report, Reference (e).
12. The vendor submitted a NM LoC with noted discrepancies. The following open TDRs were adjudicated by DISA to have a minor operational impact with a vendor submitted POA&M:
    a. The SUT does not fully support SNMP and MIBs IAW IETF Standards 58 and 62.
    b. The SUT is not fully compliant with NM call detail records formats.
    c. SUT does not support management requirements for ASAC.
13. The SUT does not support destination code controls. The SUT does not have the capability of setting the percentage of calls to be blocked to the designated destination(s). This was adjudicated by DISA to have a minor operational impact.

Table 2-6. SUT CR and FR Status (continued)

LEGEND:

AEI	Assured Services End Instrument
APL	Approved Products List
ASAC	Assured Services Admission Control
AS	Assured Services
ASD/NII	Assistant Secretary of Defense for Networks and Information Integration
AS-SIP	Assured Services Session Initiation Protocol
CAS	Channel Associated Signaling
CCA	Call Connection Agent
CCS7	Common Channel Signaling Number 7
CR	Capability Requirement
CM	Configuration Management
DISA	Defense Information Systems Agency
DoD	Department of Defense
DSCP	Differentiated Services Code Point
DSN	Defense Switched Network
EBC	Edge Boundary Controller
EMS	Element Management System
FCAPS	Fault, Configuration, Accounting, Performance, and Security
FR	Functional Requirement
IA	Information Assurance
ID	Identification
IP	Internet Protocol
IPSec	Internet Protocol Security
IPv6	Internet Protocol version 6
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
IWF	Interworking Function
LDAP	Lightweight Directory Access Protocol
LoC	Letter of Compliance
LSC	Local Session Controller
Mbps	Megabits per second
MG	Media Gateway
MGC	Media Gateway Controller
MIB	Management Information Base
NM	Network Management
NMS	Network Management System
PEI	Proprietary End Instrument
POA&M	Plan of Action and Milestones
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
RFC	Request for Comment
SG	Signaling Gateway
SIP	Session Initiation Protocol
SNMP	Simple Network Management Protocol
SNMPv2	Simple Network Management Protocol version 2
SNMPv3	Simple Network Management Protocol version 3
SS	Softswitch
SS7	Signaling System 7
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
TDM	Time Division Multiplexing
TDR	Test Discrepancy Report
UC	Unified Capabilities
UCR	Unified Capabilities Requirements
VoIP	Voice over Internet Protocol
VVoIP	Voice and Video over Internet Protocol
WAN	Wide Area Network
WWNDP	World Wide Numbering and Dialing Plan

a. Assured Services Product Features and Capabilities.

(1) **Differentiated Services Code Point** (**DSCP) Packet Marking**. As part of the session setup process, the WAN SS controls what DSCP to use in the subsequent session media stream packets. The exact DSCP method used shall comply with UCR 2008 Change 1, Section 5.3.3.3.2. The SUT met all DSCP Packet Marking requirements for IPv4. The SUT AS-SIP trunking interfaces were not tested for IPv6 only the integrated LSC was tested and met IPv6 DSCP Marking requirements (see paragraph 11.2. p).

(2) **Assured Services Admission Control** (**ASAC) – Open Loop**. The WAN SS must meet the ASAC requirements for the LSC and the WAN SS. In the execution of ASAC, certain procedures need to be followed, such as (a) actions to be taken if a precedence session request cannot be completed because existing sessions are at equal or higher precedence, or (b) tones to be generated when a session is preempted. The SUT met all ASAC requirements.

(3) **Signaling Protocols**. The WAN SS must use appropriate signaling for specific trunk types. The control/management protocol between the PEI and the LSC is, in general, proprietary. The control/management protocol between the AEI and the

LSC is AS-SIP as specified in UCR 2008 Change 1, Section 5.3.4, AS-SIP Requirements, of this document. The signaling protocol used on UC IP trunks is AS-SIP as specified in UCR 2008 Change 1, Section 5.3.4, AS-SIP Requirements. The MG3K MG within the WAN SS uses ANSI T1.619a ISDN PRI NI2 (Assured Services), ANSI T1.607 ISDN PRI NI2 (PSTN), and ETSI E1 ISDN PRI Q.931 (PSTN) interfaces. The ETSI E1 ISDN PRI Q.931 interfaces is a copper interface. The ANSI T1.619a ISDN PRI NI2 and ANSI T1.607 ISDN PRI NI2 interfaces are certified for joint use via copper and OC3 VT1.5 via the MG. The SUT met all Signaling Protocol requirements.

(4) **Signaling Performance**. The SUT met all signaling performance requirements.

b. Registration, Authentication, and Failover. Each WAN SS shall be configured with knowledge of each pair of WAN SS's that act as backups for each other. Each WAN SS shall be able to failover to the backup WAN SS when another WAN SS loses connectivity. Each WAN SS shall be able to failback to the original WAN SS upon recovery. WAN SS failover was not tested due to the limited number of WAN SS's available at the time of test. This requirement was met via vendor LoC.

c. Product Physical, Quality, and Environmental Factors. The Assured Services subsystem shall have a hardware/software availability of 0.99999 (non-availability of no more than 5 minutes per year). This requirement was met via vendor LoC.

d. SS Requirements.

(1) **LSC Monitors Primary SS for Status**. Per the requirement in UCR Change 1, paragraph 5.3.2.3.2.2 when a properly functioning primary SS receives the OPTIONS request from a served LSC, the primary SS shall respond with a 200 OK response that includes the Accept header and the Supported header. The SUT partially met this requirement through a vendor submitted NM LoC with noted discrepancies. Those discrepancies are:

- The SUT does not fully support SNMP and MIBs IAW Internet Engineering Task Force (IETF) Standards 58 and 62.
- The SUT is not fully compliant with NM call detail records formats.
- The SUT does not support management requirements for ASAC.

The open TDRs were adjudicated by DISA to have a minor operational impact with a vendor submitted POA&M.

(2) **SS Monitors every other SS in the network**. Per the requirement in UCR 2008 Change 1, paragraph 5.3.2.3.2.5, each SS shall send an OPTIONS request to every other SS on a "standard" configurable periodic time interval (default equals 45 seconds; minimum time interval equals 35 seconds). In each OPTIONS request, the Request-Uniform Resource Identifier (URI) identifies the destination SS (the Request-

URI does not have a userinfo part). The OPTIONS requests shall include a route set comprised of two Route Headers, where the first Route Header is the SIP URI for the Edge Boundary Controller (EBC) of the SS originating the OPTIONS request, and the second Route Header is the SIP URI for the EBC serving the destination SS. The SUT partially met this requirement through a vendor submitted NM LoC with noted discrepancies. Those discrepancies are:

- The SUT does not fully support Simple Network Management Protocol (SNMP) and Management Information Bases (MIB) IAW IETF Standards 58 and 62.
- The SUT is not fully compliant with NM call detail records formats.
- The SUT does not support management requirements for ASAC.

The open Test Discrepancy Reports (TDR) were adjudicated by DISA to have a minor operational impact with a vendor submitted POA&M.

Whenever an originating SS sends an INVITE request to another SS and receives either a 408 (Request Time-Out), 503 (Service Unavailable), or 504 (Server Time-Out) response and the originating SS is not already awaiting a response to a pending OPTIONS request to the other SS, then the originating SS shall send an OPTIONS request with a Request-URI identifying the SS. The OPTIONS request shall include a route set comprised of two Route Headers, where the first Route Header is the SIP URI for the EBC of the SS originating the OPTIONS request, and the second Route Header is the SIP URI for the EBC serving the destination SS. The SUT partially met this requirement through a vendor submitted NM LoC with noted discrepancies. Those discrepancies are:

- The SUT does not fully support SNMP and MIBs IAW IETF Standards 58 and 62.
- The SUT is not fully compliant with NM call detail records formats.
- The SUT does not support management requirements for ASAC.

The open TDRs were adjudicated by DISA to have a minor operational impact with a vendor submitted POA&M.

When a properly functioning SS receives the OPTIONS request, the SS shall respond with a 200 OK response that includes the accept header and the supported header. The SUT partially met this requirement through a vendor submitted NM LoC with noted discrepancies. Those discrepancies are:

- The SUT does not fully support SNMP and MIBs IAW IETF Standards 58 and 62.
- The SUT is not fully compliant with NM call detail records formats.
- The SUT does not support management requirements for ASAC.

The open TDRs were adjudicated by DISA to have a minor operational impact with a vendor submitted POA&M.

(3) **ASAC Requirements.** Per the requirement in UCR 2008 Change 1, paragraph 5.3.2.18.2, the ASAC must provide the separate counts for voice and video, in five-minute intervals. The MFSS and WAN SS ASAC must provide these counts for each of the subtended LSCs under its control, while the LSC is only to provide these counts for the PEIs/AEIs that it controls. The ASAC reporting parameters are shown in Table 5.3.2.18-4, ASAC Reporting Parameters. The SUT met this requirement.

(4) **Call Data**. Per the requirement in UCR 2008 Change 1, Section 5.3.2.19.2.1, regardless of the recording format that is chosen, there is information that is important for proper billing and accounting. Also, there is information that may be necessary for one type of call, but may not be necessary for another type of call. This requirement lists the call data that are needed in the recording. The SUT met this requirement.

(5) **Destination Code Controls**. Per the requirement in UCR 2008 Change 1, paragraph 5.3.2.17.3.4.2.7, within the IP environment, Destination Code Control functionality is applied at the LSC, WAN SS or MFSS to prevent or limit the number of calls (session requests) to reach a specific destination. Destination code controls are applied to reduce calls to a specific area or location that has been temporarily designated as "difficult to reach" due to several circumstances. The SUT partially met this requirement through a vendor submitted LoC with noted discrepancies. The SUT cannot block calls to a specific destination code based on a percentage. This discrepancy was adjudicated by DISA to have a minor operational impact with a vendor submitted POA&M.

(6) **Directionalization**. Per the requirement in UCR 2008 Change 1, paragraph 5.3.2.17.3.4.2.10, directionalization is intended to control the relative volume of call initiation from on base to off base, or vice versa (i.e., control the sourcing direction). This requirement stems from the Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01C, Appendix A. The SUT did not meet this requirement; however, UCR 2008 Change 2 has modified this requirement as conditional for a WAN SS.

(7) **Call budget**. Per the requirement in UCR paragraph 5.3.2.17.3.4.2.13, Setting the Call Budgets on the MFSS, WAN SS, and LSC involves setting the maximum number of calls (voice and video) that may be in service at one time within, and/or to/from a local service area (i.e., military installation). Two call budget actions or functions can be performed: Setting the total call budget, and designating all or part of the call budget as inbound (local destination) and/or outbound (i.e., local origination) (to be able to implement an IP equivalent of directionalization). The default for the directionalization is no designation (i.e., calls can be inbound or outbound in any combination).

The above defined call budget actions for the MFSS and WAN SS will be applied to the WAN-level ASAC. The WAN-level ASAC must be able to account for each subtended LSC under its control. Therefore, the MFSS and WAN SS ASAC must be able to set call budgets for multiple LSC locations via the Voice and Video over Internet Protocol (VVoIP) EMS and local EMS access points. The SUT met this requirement with the exception of directionalization which has changed to a conditional requirement in UCR 2008 Change 1.

(8) **Quality of Service**. Per the requirement in UCR 2008 Change 1, paragraph 5.3.2.19.2.1.1, the product shall provide a voice quality record at the completion of each voice session. The voice quality record shall be included in the CDR that the LSC, MFSS, or WAN SS generates for that session, and shall conform to the E-Model, as described in TIA TSB-116-A, and ITU-T Recommendation G.107. The voice quality record shall contain the calculated R-Factor for the Voice session per TIA TSB-116-A. The allowable error for the voice quality calculations shall be ±3 in accordance with TIA TSB-116-A. The SUT does not provide any end instruments and; therefore, is not required to provide this. The optional LSC component does offer end instruments and the interoperability certification letter and test summary report for the LSC is listed on the UC APL reference (g).

(9) **AS-SIP Signaling Appliance.** When an AS-SIP signaling appliance that is implemented as a SIP proxy receives a SIP request message, 2xx response, or 18x response, then the AS-SIP signaling appliance must add a Record Route header whereby the user info of the SIP URI is a unique identifier for the AS-SIP signaling appliance and an IP address is used for the host name. This requirement was met by the SUT.

Interworking AS-SIP signaling appliances MUST be in adherence with the enumerated RFCs, be capable of generating, receiving, and processing the SIP header listed in UCR 2008 Change 1, paragraph 5.3.4.16.3.1. This requirement was met by the SUT.

e. Call Connection Agent (CCA) Requirements.

(1) **CCA Interworking Function (IWF) Component.** The role of the IWF within the CCA is to support all the VoIP and TDM signaling protocols that the WAN SS supports for EIs (optional with LSC), Media Gateways (MG), and EBCs, and to interwork all these various signaling protocols with one another. The SUT met all CCA IWF requirements for the following interfaces: T1 ISDN PRI for both ANSI T1.619a (Assured Services) and ANSI T1.607 (PSTN only) with NI-2, and ETSI E1 ISDN PRI Q.931 (PSTN only).

(2) **CCA MG Controller (MGC) Component.** The MGC within the CCA must control all MGs within the LSC or MFSS, control all trunks within each MG, control all signaling and media streams on each trunk within each MG, accept IP-encapsulated signaling streams from an Serial Gateway (SG) or MG, and use either ITU-T recommendation H.248 or a supplier-proprietary protocol to accomplish these controls. The SUT met all CCA MGC requirements.

(3) **SG Component.** The role of the CCA with respect to the SG is to control all SGs within the network appliance, and to control all signaling links (DoD CCS7) within each SG. The SG is conditional for a WAN SS and was not tested on the SUT.

(4) **CCA-IWF Support for AS-SIP.** The CCA IWF shall support the AS-SIP protocol consistent with the detailed AS-SIP protocol requirements in UCR 2008 Change 1, Section 5.3.4, AS-SIP Requirements. The SUT met all requirements for CCA-IWF support for AS-SIP for the following interfaces: T1 ISDN PRI for both ANSI T1.619a (Assured Services) and ANSI T1.607 (PSTN only) with NI-2, and ETSI E1 ISDN PRI Q.931 (PSTN only).

(5) **CCA-IWF Support for SS7.** CCA-IWF support for SS7 is a conditional requirement for WAN SS; the SUT does not offer this interface.

(6) **CCA-IWF Support for PRI, via MG.** The CCA IWF shall support the U.S./National ISDN version of the ISDN PRI protocol. The SUT met all requirements for CCA-IWF support for the following interfaces: T1 ISDN PRI for both ANSI T1.619a (Assured Services) and ANSI T1.607 (PSTN only) with NI-2, and ETSI E1 ISDN PRI Q.931 (PSTN only).

(7) **CCA-IWF Support for Channel Associated Signaling (CAS) Trunks via MG.** Support for CAS is a conditional requirement for a WAN SS; the SUT does not offer this interface.

(8) **CCA-IWF Support for VoIP and TDM Protocol Interworking.** The role of the IWF within the CCA is to support all the VoIP and TDM signaling protocols that the appliance supports for PEIs, AEIs, MGs, and EBCs, and interwork all these various signaling protocols with one another. The SUT met all requirements for CCA-IWF Support for VoIP and TDM Protocol Interworking for the following interfaces: T1 ISDN PRI for both ANSI T1.619a (Assured Services) and ANSI T1.607 (PSTN only) with NI-2, and ETSI E1 ISDN PRI Q.931 (PSTN only).

(9) **CCA Preservation of Call Ringing State during Failure Conditions.** The CCA in the LSC, MFSS, and WAN SS shall not allow AS-SIP sessions that have reached the ringing state (i.e., an AS-SIP 180 (Ringing) message or 183 (Session Progress) has been sent from the called party to the calling party, and the calling party is receiving an audible ringing tone) to fail when an internal failure occurs within the CCA. This requirement was not tested. This requirement represents a new UCR requirement (Jan 2010) at the time of the APL interoperability testing and therefore

compliance is not mandatory at that time, based on allowance of an 18-month development cycle for new requirements.

(10) **CCA Interactions with Transport Interface Functions.** The CCA interacts with Transport Interface functions by using them to communicate with PEIs, AEIs, the EBC, the MGs, and the SG over the ASLAN. The SUT met all requirements for CCA interactions with Transport Interface Functions exception of AEIs (on optional LSC). No AEIs were tested.

(11) **CCA Interactions with the EBC.** The CCA interacts with the EBC by directing AS-SIP signaling packets to it (for signaling messages destined for a WAN SS) and by accepting AS-SIP signaling packets from it (for signaling messages directed to the LSC from an WAN SS). The SUT met all requirements for CCA interactions with the EBC.

(12) **CCA Support for Admission Control.** The CCA interacts with the ASAC component of the LSC and WAS SS to perform specific functions related to ASAC, such as counting internal, outgoing, and incoming calls; managing separate call budgets for VoIP and Video over IP calls; and providing preemption. The SUT met all requirements for CCA support for Admission Control.

(13) **CCA Support for UFS.** The UFS Server is responsible for providing features and services to VoIP and Video PEIs/AEIs on an LSC or MFSS, where the CCA alone cannot provide the feature or service. The SUT met all requirements for CCA Support for UFS for PEIs with the optional LSC.

(14) **CCA Support for IA.** The IA function within the appliance ensures that end users, PEIs, AEIs, MGs, SGs, and EBCs that use the appliance are all properly authenticated and authorized by the appliance. The IA function ensures that Voice and Video signaling streams that traverse the appliance and its ASLAN are encrypted properly SIP/TLS. IA requirements are tested separately; see paragraph 11.3.

(15) **CCA Support for AS Voice and Video.** The CCA in the WAN SS or LSC needs to interact with VoIP PEIs and AEIs served by that WAN SS or LSC. The VoIP interface between the PEI and the WAN SS or LSC is left up to the network appliance supplier. The VoIP interface between the AEI and the WAN SS or LSC is AS-SIP. The SUT met all requirements for CCA support for AS Voice and Video.

(16) **CCA Interactions with Service Control Functions.** The CCA shall support the ability to remove VoIP and Video sessions and session requests from the media server so the CCA can continue with necessary session processing once the media server has completed its functions. The SUT met all requirements for CCA Interactions with Service Control Functions.

(17) **CCA Interworking between AS-SIP and CCS7.** Interworking is performed at a node with CCA (SIP/CCS7 IWF) functionality that processes/interworks incoming CCS7 messages to outgoing AS-SIP messages, and similarly, incoming AS-SIP messages to outgoing CCS7 messages. This is a conditional requirement for a WAN SS and is not supported by the SUT.

f. MG Requirements.

(1) **Role of MG in the WAN SS.** The MG supports interconnection of VoIP, Facsimile over IP (FoIP), and Modem over IP (MoIP) media streams with the SS media server, which provides tones and announcements for WAN SS calls and features. To support inter-enclave MoIP and FoIP, the WAN SS must meet ITU-T V.150.1 requirements. The SUT did not meet this requirement; however, the V.150.1 requirement was a new UCR requirement (Jan 2010) at the time of the APL interoperability testing and therefore compliance is not mandatory at that time, based on allowance of an 18-month development cycle for new requirements.

(2) **MG Support for ASAC.** The MG assists the CCA in performing ASAC (i.e., call preemption based on per-call precedence levels) for outgoing TDM calls at MGs and for incoming TDM calls at MGs. The SUT met all requirements for MG Support for ASAC.

(3) **MG and IA Functions.** The IA function within the appliance ensures that end users, PEIs, AEIs, MGs, SGs, and EBCs that use the appliance are all properly authenticated by the appliance. The IA function also ensures that VoIP signaling streams and media streams that traverse the appliance and its ASLAN are properly encrypted, using SIP/TLS and Secure Real-Time Transport Protocol (SRTP), respectively. IA requirements are tested separately; see paragraph 11.3.

(4) **MG Interaction with Service Control Function.** The MG is responsible for routing individual VoIP, FoIP, and MoIP media streams to the media server when instructed to do so by the CCA/MGC. When instructed to do so by the CCA/MGC, the MG is responsible for removing individual VoIP, FoIP, and MoIP media streams from the media server, and for either disconnecting them entirely, or routing them on to other LSC end users (e.g., VoIP or video EIs). The SUT met all requirements for MG Interaction with Service Control Function except for V.150.1. The V.150.1 requirement was a new UCR requirement (Jan 2010) at the time of the APL interoperability testing and therefore compliance is not mandatory at that time, based on allowance of an 18-month development cycle for new requirements.

(5) **MG Interactions with IP Transport Interface Functions**. The Transport Interface functions in the WAN SS provide interface and connectivity functions with the ASLAN and its IP packet transport network. The SUT met all requirements for MG Interactions with IP Transport Interface Functions.

(6) **MG-EBC interactions**. The MG interacts with the EBC by sending SRTP

media streams to it (for call media destined for a PEI, AEI, or MG that is served by another appliance outside the WAN SS), or by accepting SRTP media streams from it (for call media arriving from a PEI, AEI, or MG that is served by another appliance outside the WAN SS). The SUT met all requirements for MG-EBC interactions with PEIs.

(7) **MG IP-Based PSTN Interface Requirements.** Voice and Video over IP interfaces from the UC network to the PSTN have not been defined. Interfaces from an LSC or WAN SS to the PSTN will be via an MG with TDM interfaces as specified in UCR 2008 Change 1, Section 5.2, Circuit-Switched Capabilities and Features.

(8) **MG Support for User Features and Services.** The MG shall support the operation of features for VoIP and Video end users, consistent with the operation of this feature on analog and ISDN lines in DoD TDM switches today. The SUT met all requirements for MG Support for User Features and Services.

(9) **MG Interface to TDM network elements in DoD Networks.** Each appliance MG shall support TDM trunk groups that can interconnect with the following devices in DoD networks, in the United States and worldwide: PBXs, SMEOs, EOs, and MFSS. The SUT met all requirements for MG Interface to TDM devices in DoD Networks.

(10) **MG Interface to TDM Allied and Coalition.** The appliance suppliers should support TDM trunk groups on their MG product that can interconnect with NEs in U.S. allied and coalition partner networks worldwide. This requirement is conditional and was not tested.

(11) **MG Interface to TDM PSTN in US.** Each appliance MG shall support TDM trunk groups that can interconnect with network elements (NE) in the PSTN in the United States. The SUT met all requirements for MG Interface to TDM PSTN in the US using ANSI T1. 607 ISDN PRI NI-2.

(12) **MG Interfaces to TDM PSTN Outside the Continental United States (OCONUS).** The appliance supplier (i.e., LSC or WAN SS supplier) should support TDM trunk groups on its MG product that can interconnect with NEs in foreign country PTT networks (OCONUS) worldwide. The SUT met all requirements for MG Interface to TDM PSTN using ETSI E1 ISDN PRI Q.931.

(13) **MG Support for CCS7.** The MG shall support TDM trunk groups that are controlled by a separate CCA-to-SG signaling link that carries DoD CCS7 protocol. The MG shall support these TDM trunk groups, and the SG shall support DoD CCS7 signaling. This requirement is conditional for a WAN SS and is not supported by the SUT.

(14) **MG Support for ISDN PRI Trunks.** The MG shall support ISDN PRI trunk groups that carry the U.S./National ISDN version of the ISDN PRI protocol. The

SUT met all requirements for MG Support with the following interfaces: T1 ISDN PRI for both ANSI T1.619a (Assured Services) and ANSI T1.607 (PSTN only) with NI-2.

(15) **MG Support for CAS Trunks.** The MG shall support CAS trunk groups that carry the U.S. version of the CAS protocol. CAS is a conditional requirement for a WAN SS. This interface is not offered by the SUT.

(16) **MG Echo Cancellation.** The MG shall provide an EC capability with an echo path capacity (echo tail length) of at least 64 ms. The MG shall provide echo cancellation for voice, G3 Fax, and VBD modem fax calls. Each MG EC shall be equipped with an "echo canceller disabling signal" tone detector. The SUT met all requirements for MG Echo Cancellation.

(17) **MG Clock Timing.** The MG shall derive its clock timing from a designated T1 or PRI interface. The SUT met all MG Clock Timing requirements.

(18) **MG V.150.1.** When the MG uses V.150.1 inband signaling to transition between audio, FoIP, modem relay, or VBD states or modes, the MG shall continue to use the established session's protocol (e.g., decimal 17 for UDP) and port numbers so that the transition is transparent to the EBC. The V.150.1 requirement was a new UCR requirement (Jan 2010) at the time of the APL interoperability testing and therefore compliance is not mandatory at that time, based on allowance of an 18-month development cycle for new requirements.

(19) **MG Preservation of Call Ringing during Failure.** The LSC MG, MFSS MG, and WAN SS MG shall not allow AS-SIP sessions that have reached the ringing state (i.e., an AS-SIP 180 (Ringing) message or 183 (Session Progress) has been sent from the called party to the calling party, and the calling party is receiving an audible ringing tone) to fail when an internal failure occurs within that MG. This requirement was not tested on the SUT. This requirement represents a new UCR requirement (Jan 2010) at the time of the APL interoperability testing and therefore compliance is not mandatory at that time, based on allowance of an 18-month development cycle for new requirements.

g. World Wide Numbering and Dialing Plan (WWNDP) Requirements.

(1) **WWNDP**. The precedence level and dialed number input to the PEI or AEI shall be as specified in UCR 2008 Change 1, paragraph 5.2.3.5.1.2, Interswitch and Intraswitch Dialing. The SUT met all requirements for WWNDP.

(2) **DSN WWNDP**. WAN SSs must support DSN WWNDP and must support mapping of DSN telephone numbers to SIP URIs, following examples of DSN numbers using SIP URIs that use the syntax defined in RFC 3966. The SUT met all DSN WWNDP requirements.

h. Commercial Cost Avoidance. The SS must use a Commercial Cost Avoidance functionality to route calls from an IP EI to a PSTN E.164 number in a manner which will minimize commercial costs associated with DSN calls. This requirement was met with a Lightweight Directory Access Protocol (LDAP) routing database server which is certified under a separate IO certification letter listed on the UC APL.

i. AS-SIP Requirements.

(1) **SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs**. The WAN SS that support SIP EIs MUST comply with the differentiated set of requirements defined for SIP EIs if they serve SIP EIs, and WAN SS MUST comply with the differentiated set of requirements defined for H.323 and/or vendor-proprietary EIs if they serve H.323 and/or vendor-proprietary EIs. This requirement was not tested on the SUT. This requirement represents a new UCR requirement (Jan 2010) at the time of the APL interoperability testing and therefore compliance is not mandatory at that time, based on allowance of an 18-month development cycle for new requirements.

(2) **SIP Session Keep-Alive Timer**. The AS-SIP signaling appliances MUST support the keep-alive mechanism for SIP sessions in accordance with RFC 4028. The SUT met all keep-alive timer requirements.

(3) **Session Description Protocol (SDP)**. A session description consists of a session-level description (details that apply to the whole session and all media streams) and optionally several media-level descriptions (details that apply to a single media stream). The SS must support SDP in accordance with RFC 2327. The SUT met all SDP requirements.

(4) **Precedence and Preemption**. The WAN SS must meet the detailed requirements for the execution of preemption and the handling of precedence information as defined in paragraph 5.3.4.2.10 of UCR 2008 Change 1. The SUT met all precedence and preemption requirements.

(5) **Video Telephony – General Rules**. Video calls must meet the detailed requirements for video telephony messaging as defined in paragraph 5.3.4.12 of UCR 2008 Change 1. Video telephony requirements were tested and met by the SUT using softphone PC Clients.

(6) **Calling Services**. The SS must meet AS-SIP call flow requirements for calling services features as defined in paragraph 5.3.4.13 of UCR 2008 Change 1. The SUT met all calling services call flow requirements.

(7) **SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances**. This specification uses SIP translation for converting between ISUP signaling and AS-SIP signaling but does not use SIP encapsulation of ISUP. This

requirement applies to translations between AS-SIP and CCS7. This is a conditional requirement for WAN SS and is not offered by the SUT.

(8) **SIP Requirements for Interworking AS-SIP Signaling Appliances**. Interworking AS-SIP signaling appliances MUST comply with UCR 2008 paragraph 5.3.4.7.1, AS-SIP Signaling Appliances and AS-SIP EIs, as well as the additional general requirements in UCR 2008 Change 1, paragraph 5.3.4.16. The SUT met all requirements for interworking AS-SIP signaling appliances.

(9) **Keep-Alive Timer Requirements for Interworking AS-SIP Signaling Appliances**. Interworking AS-SIP signaling appliances MUST comply with UCR 2008, paragraph 5.3.4.8, SIP Session Keep-Alive Timer, as well as the additional E1 requirements listed in UCR 2008 Change 1, paragraph 5.3.4.17. The SUT met all keep-alive timer requirements for interworking AS-SIP signaling appliances.

(10) **Precedence and Preemption Extensions for Interworking AS-SIP Signaling Appliances**. The SS must meet all requirements for header fields as listed in UCR 2008 Change 1, paragraph 5.3.4.18. The SUT met all requirements for precedence and preemption extensions for interworking AS-SIP signaling appliances.

(11) **Supplementary Services**. The SS must meet call flow requirements as described in UCR 2008 Change 1, paragraph 5.3.4.19 for supplementary services. The SUT met all supplementary services requirements.

p. IPv6 Requirements.

(1) **Product Requirements.** This requirement was tested intra-enclave, and the vendor has submitted an LoC for IPv6, stating the SUT is not compliant to the following:

- RFC 4007 for IPv6 Scoped Address Architecture.
- IAD (MP 112/124): Their solution does not support IPv6 (Signaling or Media) with their MP112 and MP124 analog IADs. Per the UCR 2008 Change 1, an MFSS MG is required to support IPv6 if the packets exit the enclave and they do.
- The SESM Core supports IPv4 only for signaling and both IPv4 and IPv6 dual stack for inter-enclave. The requirement is IPv6 and IPv4 for both signaling and media.
- The Audio Codes MG3K supports IPv4 only for signaling and both IPv4 and IPv6 dual stack for media intra and inter-enclave. The requirement is IPv6 and IPv4 for both signaling and media.

q. Network Management.

(1) **General Management Requirements**. The SUT components shall each have an individual pair of Ethernet interfaces for management purposes, even in cases

where the WAN SS or LSC component contains multiple physical devices. The SUT met all general management requirements via LoC.

(2) **Requirement for Fault, Configuration, Accounting, Performance, and Security (FCAPS) Management**. The SUT must meet all general requirements for the FCAPS management functional areas as defined in UCR 2008 paragraph 5.3.2.17. The SUT does not fully comply with the UCR Change 1, paragraph 5.3.2.17.2. The SUT only supports SNMPv2 protected by IPSEC transport, while the UCR requires SNMPv3. The SUT does not download software automatically as required. The SUT can receive software electronically or physically, but does not automatically retrieve, install or rollback software updates. The SUT does not backup the software application during the backup procedure as required and this was adjudicated by DISA as having a minor operational impact with a vendor POA&M.

(3) **NM requirements of Appliance Functions**. The SUT must meet all management requirements for ASAC, CCA, and MG functions as defined in UCR 2008 Change 1, paragraph 5.3.2.18. The SUT does not fully meet the NM requirements of appliance functions. The SUT does not download software automatically as required. The SUT can receive software electronically or physically, but does not automatically retrieve, install or rollback software updates. The SUT AudioCodes EMS is reachable by the Local EMS, but not by the VVoIP EMS. ADIMSS is not managing the LSC, so this access is provided only to the Local EMS. The UCR requires the EMS to be reachable by the VVoIP EMS. No consolidation of alarms is performed by the SUT to provide only root cause alarms as required. The suppression function is not supported by the SUT as required. The SUT MG keeps history for 2 intervals of 15 minutes, instead of the required current interval and 8 hours of history data. The SUT does not support management requirements for ASAC as defined in the UCR. These discrepancies were adjudicated by DISA as having a minor operational impact with a vendor POA&M.

(4) **Accounting Management**. Accounting management identifies a set of events during which call detail information is collected. These events are call connect, call attempt, and call disconnect. When these events are detected, specific call data will be provided by the network appliances that were involved in the event. The SUT met all accounting management requirements via LoC.

11.3 Information Assurance. The IA report is published in a separate report, Reference (e).

11.4 Other. None

12. TEST AND ANALYSIS REPORT. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System 2-7 Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi.

SYSTEM FUNCTIONAL AND CAPABILITY REQUIREMENTS

The Internet Protocol Call Control products have required and conditional features and capabilities that are established by the Unified Capabilities Requirements (UCR). The System Under Test (SUT) need not provide conditional requirements. If they are provided, they must function according to the specified requirements. The detailed Functional requirements (FR) and Capability Requirements for Multi-Function SoftSwitch (MFSS), Local Session Controller (LSC), and Wide Area Network SoftSwitch (WAN SS) are listed in Table 3-1. Detailed Information Assurance (IA) requirements are included in Reference (e) and are not listed below.

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
1	As part of the session setup process, the LSC controls what DSCP to use in the subsequent session media stream packets. (two sub requirements)	5.3.2.2.1.4	Y	Y	
2	The SUT must provide the following features: Precedence Call Waiting, Call Forwarding, Call Transfer, Call Hold, Three-Way Calling, Hotline Service, and Calling Party and Called Party ID.	Table 5.3.2.2-1	Y	Y	
3	Calls to a DN that does not have any CF feature activated shall be delivered to the DN EI IAW the MLPP procedures specified in UCR 2008, Section 5.2.2 Multilevel Precedence and Preemption	5.3.2.2.2.1.1	Y	Y	
4	Call forwarding, when activated on a line DN, shall allow any terminating call at a ROUTINE DSN precedence level, to be completed to the designated destination (IAW the call forward options activated), and shall comply with the requirements as stated in Telcordia Technologies GR-217-CORE, GR-580-CORE, and GR-586-CORE.	5.3.2.2.2.1.1	Y	Y	
5	Calls to 911 shall be preempted in accordance with assured service priority rules specified in UCR 2008, Section 5.2.2, Multilevel Precedence and Preemption.	5.3.2.2.2.2.1	Y	Y	
6	The Tracing of Terminating Calls feature identifies the calling number on intraoffice and interoffice calls terminating to a specified DN. When this feature is activated, the originating DN, the terminating DN, and the time and date are printed out for each call to the specified line.	5.3.2.2.2.2.2	Y	Y	
7	The Outgoing Call Tracing feature allows the tracing of nuisance calls to a specified DN suspected of originating from a given local office. The tracing is activated when the specified DN is entered. A printout of the originating DN, and the time and date, are generated for every call to the specified DN.	5.3.2.2.2.2.3	Y	Y	
8	The Tracing of a Call in Progress feature identifies the originating DN for a call in progress. Authorized personnel entering a request that includes the specific terminating DN involved in the call activate the feature.	5.3.2.2.2.2.4	Y	Y	
9	The Tandem Call Trace feature identifies the incoming trunk of a tandem call to a specified office DN. The feature is activated by entering the specified distant office DN for a tandem call trace. A printout of the incoming trunk number and terminating DN, and the time and date, is generated for every call to the specified DN.	5.3.2.2.2.2.5	Y	Y	
10	One voice session budget unit shall be equivalent to 110 kilobits per second (kbps) of access circuit bandwidth independent of the PEI or AEI codec used. This includes ITU-T Recommendation G.711 encoding rate plus Internet Protocol Version 6 (IPv6) packet overhead plus ASLAN Ethernet overhead. IPv6 overhead, not IPv4 overhead, is used to determine bandwidth equivalents here.	5.3.2.2.2.3.1	Y	Y	
11	If the MFSS's count of an IPC is greater than or equal to the corresponding IPB, and it receives an INVITE request for a precedence session, the MFSS shall preempt a lower priority session (if such a session exists), and then proceed with processing the higher precedence session connect request.	5.3.2.2.2.3.1.2	Y		
12	If the MFSS receives a CCA-ID for which there is no entry in ASAC budget table, the SS will reject the session and generate an alarm for the EMS.	5.3.2.2.2.3.1.2	Y		
13	If necessary, the MFSS will preempt for a session request that is at precedence level FLASH OVERRIDE or FLASH and the counts equal the budgets.	5.3.2.2.2.3.2	Y		

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
14	Registration and authentication between NEs shall follow the requirements set forth in UCR 2008, Section 5.4, Information Assurance Requirements.	5.3.2.3.1	Y	Y	
15	The LSCs shall be registered to a primary and backup MFSS. In case of failure of the primary MFSS, the LSC will default to the backup MFSS.	5.3.2.3.2	Y	Y	
16	The LSC shall send an OPTIONS request with a Request-URI identifying the primary SS (the Request-URI does not have a userinfo part) on a configurable periodic time interval (default equals 45 seconds; minimum time interval equals 35 seconds). (4 sub requirements)	5.3.2.3.2.2		Y	
17	When a properly functioning primary SS receives the OPTIONS request from a served LSC, the primary SS shall respond with a 200 OK response that includes the Accept header and the Supported header.	5.3.2.3.2.2	Y		Y
18	When the LSC sends a defined configurable number of successive OPTIONS requests (default equals 2) for which there either is no response or the response is a 408 (Request Time-Out), 503 (Service Unavailable), or 504 (Server Time-Out) response, then it must failover to the secondary SS. (3 sub requirements)	5.3.2.3.2.3		Y	
19	If the LSC receives a 200 OK response to an OPTIONS request from the primary SS before the configurable number of successive failures to the OPTIONS requests (default equals 2) has been reached, then no action is taken to failover to the secondary SS.	5.3.2.3.2.3		Y	
20	Upon failover, the LSC will send OPTIONS requests to the primary SS at a failback configurable periodic time interval (default equals 60 seconds; minimum time interval equals 35 seconds). (4 sub requirements)	5.3.2.3.2.4		Y	
21	Each SS shall send an OPTIONS request to every other SS on a “standard” configurable periodic time interval (default equals 45 seconds; minimum time interval equals 35 seconds).	5.3.2.3.2.5	Y		Y
22	Whenever an originating SS sends an INVITE request to another SS and receives either a 408 (Request Time-Out), 503 (Service Unavailable), or 504 (Server Time-Out) response and the originating SS is not already awaiting a response to a pending OPTIONS request to the other SS, then the originating SS shall send an OPTIONS request with a Request-URI identifying the SS.	5.3.2.3.2.5	Y		Y
23	When a properly functioning SS receives the OPTIONS request, the SS shall respond with a 200 OK response that includes the Accept header and the Supported header.	5.3.2.3.2.5	Y		Y
24	Each MFSS (SS) shall be configured with knowledge of each pair of SSs that act as backups for each other. (7 sub requirements)	5.3.2.3.2.6	Y		Y
25	Upon failover, the SS will send OPTIONS requests to the failed SS at a “failback” configurable periodic time interval (default equals 60 seconds; minimum time interval equals 35 seconds). (5 sub requirements)	5.3.2.3.2.7	Y		Y
26	The Assured Services subsystem shall have a hardware/software availability of 0.99999 (nonavailability of no more than 5 minutes per year).	5.3.2.5.2.1	Y	Y	
27	The performance parameters associated with the ASLAN, MFSS, and LSC, when combined, shall meet the following maximum downtime requirements: • IP (10/100 Ethernet) network links – 35 minutes/year • IP subscriber – 12 minutes/year	5.3.2.5.2.2	Y	Y	
28	For these VoIP devices, the voice quality shall have a MOS of 4.0 (R-Factor equals 80) or better, as measured in accordance with the E-Model. Additionally, these devices shall not lose two or more consecutive packets in a minute and shall not lose more than seven voice packets (excluding signaling packets) in a 5-minute period.	5.3.2.5.4	Y	Y	
29	An IP voice instrument shall be designed in accordance with the acquiring activity requirements, but the following capabilities are specifically required as indicated: • **[Objective]** DoD Common Access Card (CAC) reader • **[Required]** Display calling number • **[Required]** Display precedence level of the session • **[Required]** Support for Dynamic Host Configuration Protocol (DHCP).	5.3.2.6.1	Y	Y	
30	Tones and announcements, as required in UCR 2008, Sections 5.2.4.5.2, DSN Information Signals, and Section 5.2.2.1.3, Announcements, shall be supported, except for the loss of C2 announcement.	5.3.2.6.1.1	Y	Y	

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
31	The product shall support the origination and termination of a voice session using the following codecs: • ITU-T Recommendation G.711, to include both the µ-law and A-law algorithms • ITU-T Recommendation G.723.1 • ITU-T Recommendation G.729 or G.729A • ITU-T Recommendation G.722.1	5.3.2.6.1.2	Y	Y	
32	Voice over IP PEIs or AEIs (i.e., handset, headset, and hands-free types) shall comply with TIA-810-B, November 3, 2006.	5.3.2.6.1.3	Y	Y	
33	For Fixed-to-Fixed calls, the product shall use 20 ms as the default voice sample length, and as the basis for the voice payload packet size.	5.3.2.6.1.4	Y	Y	
34	The PEI or AEI shall be capable of authenticating itself to its associated LSC and vice versa.	5.3.2.6.1.5 5.3.2.6.2.3	Y	Y	
35	Analog instruments, including secure analog EIs, analog facsimile EIs, and analog modem EIs, shall be supported by the LSC either by a TA or an Integrated Access Device (IAD) connected to an Ethernet port.	5.3.2.6.1.6	Y	Y	
36	The LSC shall meet all the requirements for PBAS/ASAC, as appropriate for VoIP and Video over IP services, as specified in UCR 2008, Section 5.2.2, Multilevel Precedence and Preemption.	5.3.2.7.2.1		Y	
37	The LSC shall support CND, as specified in UCR 2008, Section 5.2.3.5.1.8.2, Calling Number Delivery.	5.3.2.7.2.2		Y	
38	The LSC must provide signaling on the line side for local intra-enclave subscriber-to-subscriber calls, and trunk-side signaling for calls between an external enclave and a local subscriber.	5.3.2.7.2.3		Y	
39	In the event that a total loss of connectivity to the DISN WAN occurs, the LSC shall provide the following functions: • Completion of local (intra-enclave) calls • Routing of calls to the PSTN using a local MG (PRI or CAS as required by the local interface) • User look-up of local directory information	5.3.2.7.2.4		Y	
40	The LSC Management function supports functions for LSC FCAPS management and audit logs. Collectively, these functions are called FCAPS Management and Audit Logs.	5.3.2.7.2.6		Y	
41	The LSC Transport Interface functions provide interface and connectivity functions with the ASLAN and its IP packet transport network.	5.3.2.7.2.7		Y	
42	The LSC shall provide an interface to the DISA NMS. The interface consists of a 10/100-Mbps Ethernet connection	5.3.2.7.2.8		Y	
43	Periodically, the LSC shall verify the status of its registered and authenticated IP EIs, including operator (dial service attendant) consoles. The verification interval shall be configurable with the default set at 5 minutes.	5.3.2.7.2.10		Y	
44	Line-side custom features must not interfere with the Assured Services requirements.	5.3.2.7.2.11		Y	
45	During the call establishment process, the product shall be capable of preventing or detecting and stopping hair-pin routing loops over ANSI T1.619a and commercial PRI trunk groups (i.e., T1 PRI and E1 PRI) between a legacy switch (e.g., TDM EO) and an LSC	5.3.2.7.3		Y	
46	When the AS-SIP TDM Gateway receives a call request over an ISDN MLPP PRI then the AS-SIP TDM Gateway MUST map the telephony numbers received from the Q.931 SETUP message to SIP URIs	5.3.2.7.4.3.3	Y	Y	
47	The AS-SIP TDM Gateway MG MUST support the ITU-T Recommendation G.711 (µ-law and A-law) audio codec.	5.3.2.7.4.3.4	Y	Y	
48	The AS-SIP TDM Gateway MG MUST support RFC 4040 and the AS-SIP TDM Gateway MUST support the signaling for establishing the 64kbps unrestricted bearer per Section 5.3.4.7.7, 64 kbps Transparent Calls (Clear Channel).	5.3.2.7.4.3.4	Y	Y	
49	The AS-SIP TDM Gateway MG MUST support T.38 Fax Relay	5.3.2.7.4.3.4	Y	Y	
50	The AS-SIP TDM Gateway MG MUST support the SCIP-216 subset of V.150.1 Modem Relay (see Section 5.3.2.21.2, RTS SCIP Gateway Requirements) and the AS-SIP TDM Gateway MUST support the AS-SIP signaling requirements in support of modem relay	5.3.2.7.4.3.4	Y	Y	
51	The AS-SIP TDM Gateway MUST satisfy the Information Assurance requirements in Section 5.4 Information Assurance for a media gateway.	5.3.2.7.4.3.5	Y	Y	
52	The AS-SIP TDM Gateway MUST provide an interface to the DISA NMS. The interface MUST consist of a 10/100-Mbps Ethernet connection	5.3.2.7.4.3.9	Y	Y	

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
53	The AS-SIP IP Gateway MUST implement call count thresholds for voice sessions and for video sessions in order to perform Session Admission Control (SAC).	5.3.2.7.5.1.1	Y	Y	
54	The requirements for the TDM side of the MFSS are entirely the same as for the DSN MFS specified in UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features. The TDM side of the MFSS must meet these requirements.	5.3.2.8.2.1	Y		
55	MFSS shall support PRI signaling for TDM communication with other systems.	5.3.2.8.2.3	Y		
56	The TDM side of the MFSS shall support CCS7 signaling for communication with other TDM systems.	5.3.2.8.2.3	Y		
57	MFSS shall support AS-SIP signaling for IP communication with other MFSSs and LSCs.	5.3.2.8.2.3	Y		
58	The MFSS shall provide internal signaling and media conversion for calls between the TDM side and SS side of the MFSS.	5.3.2.8.2.3	Y		
59	The CCA/SG/MGC/MG complex in the SS side of the MFSS needs to interface and interact with the EO and Tandem functions in the TDM side of the MFSS.	5.3.2.8.2.4	Y		
60	The MFSS MG must support internal MG connections that interconnect the SS side of the MFSS with the EO and Tandem functions on the TDM side of the MFSS.	5.3.2.8.2.4	Y		
61	The MFSS MG shall interact with the MFSS MGC so that Internal MG connections between the SS and TDM sides of the MFSS support (1) Intra-MFSS calls between TDM EIs connected to the TDM side, and PEIs/AEIs connected to the SS side of the MFSS (2) Incoming and outgoing calls to/from systems external to the MFSS that require conversion between TDM and IP	5.3.2.8.2.4	Y		
62	When a U.S. ISDN PRI-based connection is used between the SS and TDM sides of the MFSS, the MFSS MG shall interact with the MFSS MGC so that U.S. ISDN PRI signaling (National ISDN PRI signaling with the Precedence Level IE and related MLPP IEs included) is used between the softswitch and TDM sides, and the T1.619/T1.619a version of the ISDN PRI MLPP feature operates correctly between the SS and TDM sides of the MFSS, for both VoIP-to-TDM calls and TDM-to-VoIP calls over this trunk group.	5.3.2.8.2.4	Y		
63	The SS side of the MFSS shall meet all the requirements for MLPP, as appropriate for VoIP and Video over IP services, as specified in Section 5.2.2, Multilevel Precedence and Preemption.	5.3.2.8.2.6	Y		
64	The SS side of the MFSS shall support CND as specified in UCR 2008, Section 5.2.3.5.1.8.2, Calling Number Delivery.	5.3.2.8.2.6	Y		
65	The requirements for SCS functions (i.e., CCA, IWF, MG, MGC, and SG) and NM are provided in separate sections of this document. The MFSS must meet all these requirements.	5.3.2.8.2.6	Y		
66	The CCA IWF must support AS-SIP and ISDN PRI protocols.	5.3.2.9.2.1	Y	Y	
67	The MGC within the CCA must control all MGs within the LSC or MFSS, support DoD ISDN trunks, control all signaling and media streams on each trunk within each MG, and accept IP-encapsulated singaling streams from an SG or MG.	5.3.2.9.2.2	Y	Y	
68	The CCA shall be responsible for controlling all the SGs within the MFSS and LSC.	5.3.2.9.2.3	Y	C	
69	The CCA shall be responsible for controlling each signaling link within each SG within the MFSS or LSC.	5.3.2.9.2.3	Y	C	
70	The CCA shall be responsible for controlling the DoD CCS7 signaling stream(s) within each signaling link within each SG.	5.3.2.9.2.3	Y	C	
71	Within the network appliance (i.e., MFSS and LSC), the CCA shall use either an IETF-standard set of CCS7-over-IP protocols, or a supplier-proprietary protocol to accomplish the above SG, signaling link, and signaling stream controls.	5.3.2.9.2.3	Y	C	
72	The CCA IWF shall support the AS-SIP protocol consistent with the detailed AS-SIP protocol requirements in Section 5.3.4, AS-SIP Requirements.	5.3.2.9.5.1	Y	Y	
73	The CCA IWF shall use the AS-SIP protocol on LSC-MFSS and MFSS-MFSS sessions.	5.3.2.9.5.1	Y	Y	
74	When the CCA IWF uses the AS-SIP protocol over the Access Segment between the EBC and the DISN WAN, or over the DISN WAN itself, the CCA IWF shall secure the AS-SIP protocol using TLS.	5.3.2.9.5.1	Y	Y	
75	The CCA IWF shall support the U.S./National ISDN version of the ISDN PRI protocol.	5.3.2.9.5.3	Y	Y	

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
76	The CCA IWF shall support reception of ISDN PRI messages from the MG and transmission of ISDN PRI messages to the MG.	5.3.2.9.5.3	Y	Y	
77	The CCA IWF shall be able to determine the ISDN PRI (and its D-Channel signaling link) that an incoming PRI message was received on, when processing an incoming PRI message from the MG.	5.3.2.9.5.3	Y	Y	
78	The CCA IWF shall be able to identify the ISDN PRI (and its D-Channel signaling link) that an outgoing PRI message will be sent on, when generating an outgoing PRI message to the MG.	5.3.2.9.5.3	Y	Y	
79	The CCA IWF shall be able to support multiple ISDN PRIs (and their D-Channel signaling links) at the MG, where each PRI is connected to a different PRI end point.	5.3.2.9.5.3	Y	Y	
80	The CCA IWF shall be able to differentiate between the individual ISDN PRIs (and their D-Channel signaling links) at the MG.	5.3.2.9.5.3	Y	Y	
81	The CCA IWF shall support the full set of ISDN MLPP requirements in ANSI T1.619 and ANSI T1.619a.	5.3.2.9.5.3	Y	Y	
82	The CCA IWF shall not support any of the ISDN MLPP requirements in ANSI T1.619 and ANSI T1.619a, on ISDN PRIs to TDM PBXs and switches in the U.S. PSTN.	5.3.2.9.5.3	Y	Y	
83	On ISDN PRIs from the CCA/MG to TDM PBXs and switches in allied and coalition partners (where those networks support U.S. “National ISDN” PRI), the CCA IWF shall support a DoD-user-configurable per-PRI option that allows the PRI to support or not support the ANSI T1.619/619a PRI MLPP feature on calls to and from that PRI.	5.3.2.9.5.3	Y	Y	
84	The CCA IWF shall be able to associate individual PRI configuration data with each individual PRI served by the MG and the CCA. The CCA IWF shall not require groups of PRIs served by the MG and the CCA to share “common” PRI configuration data.	5.3.2.9.5.3	Y	Y	
85	The CCA IWF shall support supplier-proprietary Voice and Video EIs and their associated proprietary EI signaling protocols.	5.3.2.9.5.5	Y	Y	
86	The CCA in the LSC, MFSS, and WAN SS shall not allow AS-SIP sessions that have reached the ringing state (i.e., an AS-SIP 180 (Ringing) message or 183 (Session Progress) has been sent from the called party to the calling party, and the calling party is receiving an audible ringing tone) to fail when an internal failure occurs within the CCA.	5.3.2.9.6	Y	Y	
87	The MFSS CCA shall be able to support MG connections between the SS side of the MFSS and the EO and Tandem functions on the TDM side of the MFSS.	5.3.2.10.1	Y		
88	The CCA shall support assignment of the following items to itself: • Only one CCA IP address (this one IP address may be implemented in the CCA as either a single logical IP address or a single physical IP address), • A CCA Fully Qualified Domain Name (FQDN) that maps to that IP address, and • A CCA SIP URI that uses that CCA FQDN as its domain name, and maps to the “SIP B2BUA” function within the CCA itself.	5.3.2.10.3	Y	Y	
89	The CCA shall support assignment of the following items to each SIP and AS-SIP PEI and AEI on the Appliance LAN: • Only one PEI or AEI IP address, • A PEI or AEI FQDN that maps to that IP address, and • A PEI or AEI SIP URI that uses that PEI or AEI FQDN as its domain name, and maps to the “SIP User Agent” function within the PEI or AEI.	5.3.2.10.3	Y	Y	
90	The CCA shall support assignment of the following items to each MG on the Appliance LAN: • Only one MG IP address (this one IP address may be implemented in the MG as either a single logical IP address or a single physical IP address), • An MG FQDN that maps to that IP address, and • An MG SIP URI that uses that MG FQDN as its domain name, and maps to the “UC Signaling and Media End Point” function within the MG.	5.3.2.10.3	Y	Y	
91	The CCA shall support assignment of the following items to each SG on the Appliance LAN: • Only one SG IP address (this one IP address may be implemented in the SG as either a single logical IP address or a single physical IP address), • An SG FQDN that maps to that IP address, and • An SG SIP URI that uses that SG FQDN as its domain name, and maps to the “UC Signaling End Point” function within the SG	5.3.2.10.3	Y	C	

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
92	The CCA shall support assignment of the following items to the EBC: • Only one EBC IP address (this one IP address may be implemented in the EBC as either a single logical IP address or a single physical IP address), • An EBC FQDN that maps to that IP address, and • An EBC SIP URI that uses that EBC FQDN as its domain name, and maps to the "SIP B2BUA" function within the EBC.	5.3.2.10.3	Y	Y	
93	When directing VoIP sessions to other network appliances providing voice and video services across the DISN, the CCA shall direct these VoIP sessions to the EBC, so that the EBC can process them before directing them to the network appliances on the DISN WAN.	5.3.2.10.4	Y	Y	
94	When accepting VoIP sessions from other network appliances on the DISN, the CCA shall accept these VoIP sessions from the EBC, because the EBC relays them from the network appliances on the DISN WAN.	5.3.2.10.4	Y	Y	
95	The LSC and MFSS CCA shall meet all the requirements in Section 5.3.2.2.2.3, ASAC – Open Loop. The LSC and MFSS CCA shall meet all the requirements in Section 5.3.4.10, Precedence and Preemption. The LSC and MFSS CCA shall meet all the requirements in Section 5.3.4.11, Policing of Call Count Thresholds.	5.3.2.10.5	Y	Y	
96	The CCA shall generate a redirecting number each time it forwards a VoIP or Video session request as part of a Call Forwarding feature.	5.3.2.10.6	Y	Y	
97	It is expected that all Assured Services products, such as LSCs and MFSSs, will support vendor-proprietary VVoIP features and capabilities, in addition to supporting the required VVoIP features and capabilities that are listed.	5.3.2.10.6	Y	Y	
98	The CCA shall relay received SIP and TLS authentication credentials and encryption key information from sending end systems (i.e., users, PEIs, AEIs, and EBCs) to the Information Assurance function to support the Information Assurance function's user, PEI, AEI, and EBC authentication capabilities, and its PEI, AEI, and EBC signaling stream encryption capabilities.	5.3.2.10.7	Y	Y	
99	The CCA MGC shall relay received H.248 and IPSec (or proprietary-protocol-equivalent) authentication credentials and encryption key information from sending end systems (i.e., MGs and SGs) to the Information Assurance function to support the Information Assurance function's MG and SG authentication capabilities, and its MG and SG signaling stream encryption capabilities.	5.3.2.10.7	Y	Y	
100	The CCA shall relay authentication credentials received in a SIP or AS-SIP REGISTER message from an PEI, AEI, or EBC to the Information Assurance function.	5.3.2.10.7	Y	Y	
101	The CCA shall relay TLS encryption key information received from a PEI or AEI to the Information Assurance function so the Information Assurance function can verify that this encryption key information can be used on the signaling streams for Voice or Video sessions to/from that PEI or AEI.	5.3.2.10.7	Y	Y	
102	The CCA shall relay TLS encryption key information received from an EBC to the Information Assurance function so the Information Assurance function can verify that this encryption key information can be used on the signaling streams for the Voice or Video sessions to/from that EBC.	5.3.2.10.7	Y	Y	
103	The CCA within the appliance shall support all Information Assurance Appliance requirements in Section 5.4, Information Assurance Requirements, which involve the appliance's SCS functions and the appliance's MGC.	5.3.2.10.7	Y	Y	
104	The CCA shall support supplier-proprietary Voice and Video EIs, using EI-CCA protocols that are proprietary to the LSC or MFSS supplier.	5.3.2.10.10	Y	Y	
105	When the CCA IWF supports AS-SIP Voice and Video AEIs, the IWF shall support these AEIs using the set of AS-SIP protocol requirements in Section 5.3.2.22, Generic AS-SIP End Instrument and Video Codec Requirements, and Section 5.3.4, AS-SIP Requirements.	5.3.2.10.10	Y	Y	
106	The Appliance CCA (i.e., LSC or MFSS) shall support both assured Voice and Video services. The CCA shall support both assured Voice and assured Video sessions, and shall support these sessions from both VoIP EIs and Video EIs, as described in UCR 2008, Section 5.3.2.10.10, CCA Interactions with End Instrument(s).	5.3.2.10.11	Y	Y	
107	The Appliance CCA shall support common procedures and protocol for VoIP and Video session control.	5.3.2.10.11	Y	Y	
108	The Appliance CCA shall support common procedures and protocol for feature control, for the features and capabilities given in Table 5.3.2.2-1, Assured Services Product Features and Capabilities.	5.3.2.10.11	Y	Y	

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
109	On calls to and from Proprietary VoIP and Proprietary Video EIs, the CCA shall use the appropriate parameters within the appliance supplier's Proprietary protocol messages to differentiate Proprietary VoIP sessions from Proprietary Video sessions.	5.3.2.10.11	Y	Y	
110	When AS-SIP EIs are supported on calls to and from AS-SIP EIs, the CCA shall use the SDP message bodies in AS-SIP INVITE, UPDATE, REFER, and ACK messages, as well as the SDP message bodies in AS-SIP 200 OK responses and earlier 1xx provisional responses, to differentiate AS-SIP Voice sessions from AS-SIP Video sessions.	5.3.2.10.11	Y	Y	
111	The CCA shall track VoIP sessions against corresponding Appliance VoIP budgets, and shall separately track Video sessions against corresponding Video budgets. The CCA shall maintain the Appliance's VoIP budgets separate from the Appliance's Video budget.	5.3.2.10.11	Y	Y	
112	As part of LSC-Level ASAC and WAN-Level ASAC Policing, the CCA shall support PBAS/ASAC for both VoIP sessions and Video sessions.	5.3.2.10.11	Y	Y	
113	The CCA shall allow an individual EI to support both VoIP and Video sessions. The CCA shall allow an individual EI to have both VoIP and Video sessions active at the same time.	5.3.2.10.11	Y	Y	
114	The CCA shall support the routing of both VoIP and Video session requests from LSCs to MFSSs, from MFSSs to LSCs, and from MFSSs to MFSSs, using AS-SIP. The CCA shall direct outgoing VoIP and Video session requests to EBCs, and shall accept incoming VoIP and Video session requests from EBCs, consistent with this LSC-to-MFSS routing, MFSS-to-LSC routing, and MFSS-to-MFSS routing.	5.3.2.10.11	Y	Y	
115	The CCA shall support the ability to remove VoIP and Video sessions and session requests from the media server so the CCA can continue with necessary session processing once the media server has completed its functions.	5.3.2.10.12	Y	Y	
116	The MG supports interconnection of VoIP, FoIP, and MoIP media streams with the following LSC functions and end-user devices: a. The LSC media server, which provides tones and announcements for LSC calls and LSC features b. AS-SIP VoIP, FoIP, and MoIP AEIs on the LSC	5.3.2.12.3.1		Y	
117	The MFSS MG shall be able to support MG trunk groups (referred to as internal MG connections) that either interconnect the SS (VoIP) side of the MFSS with the EO or Tandem functions on the TDM side of the MFSS.	5.3.2.12.3.2.1	Y		
118	On incoming call requests to a TDM trunk group, where the CCA/MGC applies a CAC Call Denial treatment to that call request, the MG shall connect the TDM called party on the incoming call request to the appropriate CAC Call Denial tone or announcement when instructed to do so by the MGC.	5.3.2.12.4.1.1	Y	Y	
119	On incoming calls or call requests to a TDM trunk group, where the CCA/MGC applies an ASAC Call Preemption treatment to that call or call request, the MG shall connect the TDM calling party on the incoming call or call request to the appropriate ASAC Call Preemption tone or announcement when instructed to do so by the MGC.	5.3.2.12.4.1.1	Y	Y	
120	On outgoing calls or call requests from a TDM trunk group, where the CCA/MGC applies an ASAC Call Preemption treatment to that call or call request, the MG shall connect the TDM called party on the outgoing call or call request to the appropriate ASAC Call Preemption tone or announcement when instructed to do so by the MGC.	5.3.2.12.4.1.1	Y	Y	
121	Each MG within an appliance shall support all the appliance requirements in Section 5.4, Information Assurance Requirements, that involve an Appliance MG.	5.3.2.12.4.2	Y	Y	
122	When instructed to do so by the MGC, the MG shall direct TDM calls and call requests to the media server.	5.3.2.12.4.3	Y	Y	
123	Since each Appliance MG is an IP endpoint on the Appliance LAN, each MG shall support assignment of the following items to itself: • Only one MG IP address (This one IP address may be implemented in the CCA as either a single logical IP address or a single physical IP address.) • An MG FQDN that maps to that IP address • An MG SIP URI that uses that MG FQDN as its domain name, and maps to a "SIP User Agent" function within the MG.	5.3.2.12.4.4	Y	Y	

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
124	The MG shall interact with the Transport Interface functions in the appliances when the MG uses the native LAN protocols, IP, and UDP to exchange SRTP media streams with PEIs, AEIs, other MGs, and the EBC over the Appliance LAN	5.3.2.12.4.4	Y	Y	
125	When sending VoIP media streams to PEIs or AEIs and MGs served by other network appliances, the MG shall direct these VoIP media streams to the EBC so the EBC can process them before sending them on to the remote PEIs or AEIs and MGs via the DISN WAN.	5.3.2.12.4.5	Y	Y	
126	When accepting VoIP media streams from PEIs or AEIs and MGs served by other network appliances, the MG shall accept these VoIP media streams from the appliance EBC, because the EBC relays them from the DISN WAN and the remote PEIs or AEIs and MGs on the DISN WAN. The MG shall recognize and act on the network-level IP addresses of the remote PEIs or AEIs and MGs, when accepting the VoIP sessions through the EBC from the DISN WAN and the remote PEIs or AEIs and MGs.	5.3.2.12.4.5	Y	Y	
127	The MG shall support the exchange of VoIP media streams with the following voice PEIs and AEIs both on the local appliance and on remote network appliances: a. Supplier-proprietary voice PEIs b. Voice SIP EIs, when the appliance supplier supports these EIs c. Voice H.323 EIs, when the appliance supplier supports these EIs d. Voice AS-SIP AEIs	5.3.2.12.4.8	Y	Y	
128	The MG shall support the operation of the following features for VoIP and Video end users, consistent with the operation of this feature on analog and ISDN lines in DoD TDM switches today: • Call Hold • Music on Hold • Call Waiting • Precedence Call Waiting • Call Forwarding Variable • Call Forwarding Busy Line • Call Forwarding No Answer • Call Transfer • Three-Way Calling • Hotline Service • Calling Party and Called Party ID (number only) • Call Pickup	5.3.2.12.4.9	Y	Y	
129	Each appliance MG shall support TDM trunk groups that can interconnect with the following NEs in DoD networks, in the United States and worldwide: • PBXs • SMEOs • EOs • MFSs Media Gateway support for these TDM trunk groups shall be identical to the support for these trunk groups in DoD TDM PBXs, EOs, Tandem switches, and MFSs today, as specified in UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features.	5.3.2.12.5	Y	Y	
130	Each appliance MG shall support TDM trunk groups that can interconnect with NEs in the PSTN in the United States, including CONUS, Alaska, Hawaii, and U.S. Caribbean and Pacific Territories. Media Gateway support for these TDM trunk groups to the U.S. PSTN shall be identical to the support for these trunk groups in DoD TDM PBXs, EOs, Tandem Switches, and MFSs today, as specified in UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features.	5.3.2.12.7	Y	Y	
131	The MG shall support foreign country ISDN PRI, where the MG handles both the media channels and the signaling channel: 1. For interconnection with a foreign country PSTN using foreign country ISDN PRI, from the country where the DoD user's B/P/C/S is located. 2. Support for ETSI PRI is required on LSC trunk groups when the LSC is used in OCONUS ETSI-compliant countries. 3. Support for ETSI PRI is required on MFSS trunk groups when the MFSS is used in OCONUS ETSI-compliant countries. 4. Support for MLPP using ISDN PRI is not required on the above trunk groups.	5.3.2.12.8	Y	Y	
132	The MG shall support ISDN PRI trunk groups that carry the U.S./National ISDN version of the ISDN PRI protocol. The MG shall support these U.S. PRI trunk groups conformant with the detailed U.S. ISDN PRI requirements.	5.3.2.12.10	Y	Y	

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
133	The MG shall support multiple U.S. PRI trunk groups based on the needs of the DoD user deploying the appliance. The MG shall allow each U.S. PRI trunk group at the MG to connect to: TDM EO and tandem components of the local MFSS; a different U.S. PSTN TDM NE (e.g., PBX, TDM switch); a different DoD TDM NE (e.g., PBX, TDM switch); or a different DoD IP NE (e.g., LSC, MFSS), based on the interconnection needs of the DoD user.	5.3.2.12.10	Y	Y	
134	The MG shall support reception of ISDN PRI messages from the CCA MGC and transmission of ISDN PRI messages to the CCA MGC.	5.3.2.12.10	Y	Y	
135	The MG shall connect to the ASLAN of the appliance using the physical layer and data link layer protocols of the ASLAN. In this case, the MG shall appear to the MGC, EBC, and appliance PEIs/AEIs as a physical layer and data link layer endpoint on a LAN switch in the ASLAN.	5.3.2.12.12.1	Y	Y	
136	The MG shall connect to the ASLAN of the appliance using the IP as a Network Layer Protocol. In this case, the MG shall appear to the MGC, EBC, and appliance PEIs/AEIs as an IP endpoint on an IP router on the ASLAN.	5.3.2.12.12.2	Y	Y	
137	The MG shall support IPv4 as a Network Layer Protocol. The MG shall also support IPv6 as a Network Layer Protocol.	5.3.2.12.12.2	Y	Y	
138	Conformant with Section 5.3.5, IPv6 Requirements, the MG shall support dual IPv4 and IPv6 stacks (i.e., support both IPv4 and IPv6 in the same IP end point) as described in RFC 4213.	5.3.2.12.12.2	Y	Y	
139	The MG shall support exchange of VoIP media streams with appliance PEIs/AEIs, other appliance MGs, and the appliance EBC (and through the appliance EBC, with other PEIs/AEIs and MGs on other network appliances) using the following IETF-defined Media Transfer Protocols: • SRTP, conformant with RFC 3711 • SRTCP, conformant with RFC 3711	5.3.2.12.12.4	Y	Y	
140	The MG shall secure all VoIP media streams exchanged with appliance PEIs/AEIs, other appliance MGs, and the appliance EBC (and through the EBC, with PEIs/AEIs and MGs on other network appliances) using SRTP and SRTCP.	5.3.2.12.12.4	Y	Y	
141	The MG shall use UDP as the underlying Transport Layer Protocol, and IP as the underlying Network Layer Protocol, when SRTP is used for media stream exchange.	5.3.2.12.12.4	Y	Y	
142	When the VoIP signaling streams contain supplier-proprietary protocol messages instead of H.248 or ISDN PRI messages, the MG shall secure the proprietary protocol message exchange with the MGC using mechanisms that are as strong as, or stronger than, the use of IPSec to secure H.248 and PRI message exchange.	5.3.2.12.12.5	Y	Y	
143	The MG shall support TDM voice streams using the following: • ITU-T 64 kbps G.711 µ-law PCM over digital trunks • ITU-T 64 kbps G.711 A-law PCM over digital trunks • North American 56 kbps G.711 µ-law PCM over digital trunks • North American analog voice transmission over analog trunks on TDM trunk groups on the TDM side of the MG	5.3.2.12.12.6.5	Y	Y	
144	The MG shall convert between North American 56 kbps G.711 µ-law PCM and ITU-T 64 kbps G.711 µ-law PCM in cases where North American 56 kbps TDM voice trunks are used on the TDM side of the MG.	5.3.2.12.12.6.5	Y	Y	
145	The MG shall convert between North American analog voice transmission and ITU-T 64 kbps G.711 µ-law PCM in cases where North American analog voice trunks are used on the TDM side of the MG.	5.3.2.12.12.6.5	Y	Y	
146	The MG shall support uncompressed, packetized VoIP streams using ITU-T Recommendation G.711 µ-law PCM and ITU-T Recommendation G.711 A-law PCM (ITU-T Recommendation G.711, November 1998, plus Appendix I, September 1999, and Appendix II, September 2000) over the IP network on the VoIP side of the MG.	5.3.2.12.12.6.5.1	Y	Y	
147	The MG shall packetize/depacketize G.711 media streams received or sent between its TDM side and its VoIP side.	5.3.2.12.12.6.5.1	Y	Y	
148	The MG shall transport each packetized G.711 VoIP stream to and from the destination local PEI, local AEI, local MG, remote PEI (via an EBC), remote AEI (via an EBC), or remote MG (via an EBC) using SRTP, UDP, and IP protocol layers on the VoIP side of the MG.	5.3.2.12.12.6.5.1	Y	Y	
149	The MG shall support the use of uncompressed, packetized G.711 µ-law and A-law VoIP media streams for both Fixed and Deployable applications.	5.3.2.12.12.6.5.1	Y	Y	
150	The MG shall provide an EC capability with an echo path capacity (echo tail length) of at least 64 ms.	5.3.2.12.13.2.2	Y	Y	

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
151	The MG shall provide echo cancellation for voice, G3 Fax, and VBD modem fax calls.	5.3.2.12.13.2.2	Y	Y	
152	Each MG EC shall be equipped with an “echo canceller disabling signal” tone detector. This tone detector shall detect and respond to an in-band EC disabling signal from an end user’s G3 Fax or VBD modem device. The EC disabling signal detected shall consist of a 2100-Hz tone with periodic phase reversals inserted in that tone.	5.3.2.12.13.2.2	Y	Y	
153	The MG tone detector/EC disabler shall detect the “echo canceller disabling signal” and disable the MG EC when, and only when, that signal is present for G3 Fax or VBD modem.	5.3.2.12.13.2.2	Y	Y	
154	The MG shall derive its clock timing from a designated T1 or PRI interface.	5.3.2.12.14	Y	Y	
155	The MGC within the CCA shall be responsible for controlling all the MGs within the LSC or MFSS.	5.3.2.12.15	Y	Y	
156	The MGC within the CCA shall be responsible for controlling all the trunks (i.e., DoD CCS7, PRI, or CAS) within each MG within the LSC or MFSS.	5.3.2.12.15	Y	Y	
157	The MGC within the CCA shall be responsible for controlling all media streams on each trunk within each MG.	5.3.2.12.15	Y	Y	
158	The MGC within the CCA shall accept IP signaling streams from an MG, conveying received PRI or CAS trunk signaling. The MGC shall return IP signaling streams to the MG accordingly, for conversion to transmitted PRI or CAS trunk signaling.	5.3.2.12.15	Y	Y	
159	Within the appliance (i.e., LSC or MFSS), the MGC shall use either ITU-T Recommendation H.248 (Gateway Control Protocol Version 3) or a supplier-proprietary protocol to accomplish the MG, trunk, and media stream controls described previously.	5.3.2.12.15	Y	Y	
160	Whenever the MG uses ITU-T Recommendation V.150.1, the following applies: ITU-T Recommendation V.150.1 provides for three states: audio, VBD, and modem relay. After call setup, inband signaling may be used to transition from one state to another. In addition, V.150.1 provides for the transition to FoIP using Fax Relay per ITU-T Recommendation T.38.	5.3.2.12.16	Y	Y	
161	The LSC MG, MFSS MG, and WAN SS MG shall not allow AS-SIP sessions that have reached the ringing state (i.e., an AS-SIP 180 (Ringing) message or 183 (Session Progress) has been sent from the called party to the calling party, and the calling party is receiving an audible ringing tone) to fail when an internal failure occurs within that MG.	5.3.2.12.17	Y	Y	
162	The precedence level and dialed number input to the PEI or AEI shall be as specified in UCR 2008, Section 5.2.3.5.1.2, Interswitch and Intraswitch Dialing.	5.3.2.16	Y	Y	
163	The DSN Worldwide Numbering and Dialing Plan will be used as the addressing schema within the current DSN and its migration into the SIP environment.	5.3.2.16.1	Y	Y	
164	The CCA shall allow session requests from LSC, MFSS EIs, other appliances, and MFSS MGs to contain • Called addresses including DSN numbers from the DSN numbering plan • Called addresses including E.164 numbers from the E.164 numbering plan	5.3.2.16.1	Y	Y	
165	When a session request’s called address includes a DSN number from the DSN numbering plan, the CCA shall determine whether the called DSN number is local to the LSC or MFSS, or external to the LSC or MFSS.	5.3.2.16.1	Y	Y	
166	When a session request’s called address includes an E.164 number from the E.164 numbering plan, the CCA shall determine whether the called E.164 number is local to the LSC or MFSS, or external to the LSC or MFSS.	5.3.2.16.1	Y	Y	
167	The CCA shall allow each VoIP and Video PEI and AEI served by an LSC or MFSS to have both a DSN number assigned and an E.164 number assigned.	5.3.2.16.1.1	Y	Y	
168	For VoIP and Video PEIs or AEIs that have both a DSN number and an E.164 number assigned, the CCA shall be able to match each PEI’s or AEI’s DSN number with its E.164 number, and to match each PEI’s or AEI’s E.164 number with its DSN number.	5.3.2.16.1.1	Y	Y	
169	The CCA shall be able to distinguish DSN called numbers from E.164 called numbers when processing VoIP and Video session requests from PEIs, AEIs, EBCs, MG line cards, and MG trunk groups.	5.3.2.16.1.2	Y	Y	
170	The CCA shall be able to distinguish local [DSN or E.164] called numbers from external [DSN or E.164] called numbers when processing VoIP and Video session requests from PEIs, AEIs, EBCs, MG line cards, and MG trunk groups.	5.3.2.16.1.2	Y	Y	

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
171	The MFSS or LSC is only required to support one network FQDN for use with SIP URI domain names: “uc.mil” if that appliance is used for SBU traffic, and “cuc.mil” if that appliance is used for classified traffic.	5.3.2.16.1.4.1	Y	Y	
172	The MFSS or LSC is required to ensure that all AS-SIP session requests entering or leaving that appliance use the network FQDN of that appliance (i.e., “uc.mil” for SBU traffic, or “cuc.mil” for Classified traffic) as the domain name in called SIP URIs.	5.3.2.16.1.4.1	Y	Y	
173	All voice systems, TDM or IP technology-based, must contain subscriber assignment information.	5.3.2.16.1.5	Y	Y	
174	Use of the Commercial Cost Avoidance functionality shall be an optional application that can be configured (i.e., enabled and disabled) on each RTS LSC.	5.3.2.23		Y	
175	The LSC shall be able to query the DISN RTS Routing Database on “99 dialed PSTN number” call requests from LSC end users. When the database responds to this query with a DSN number that matches the dialed PSTN number, the LSC shall route the call request over the appropriate IP (AS-SIP) or TDM (e.g., T1.619A PRI) path, using the DSN number returned by the database. When the database responds with a “number not found” indication, the LSC shall route the call request to the local TDM PSTN trunk group (PRI or CAS) on the LSC’s MG, using the originally dialed commercial number.	5.3.2.23		Y	
176	The query-response interface between the LSC and the RTS Routing Database shall be LDAP Version 3 (v3) over TLS over IP. This LDAPv3 interface shall be compliant with RFC 4510.	5.3.2.23		Y	
177	The encoding of the LDAPv3 messages and data schema used on the DB query interface between the LSC and the RTS Routing Database shall follow the BER of ASN.1, consistent with Section 5.1, Protocol Encoding, of RFC 4511.	5.3.2.23		Y	
178	The DB query interface between the LSC and the RTS Routing Database shall traverse the data firewalls (and not the RTS EBC firewalls) at both the LSC and RTS Routing Database sites.	5.3.2.23		Y	
179	After transmitting a Commercial Cost Avoidance query to the Database, the LSC shall start a “Commercial Cost Avoidance Query Response” timer awaiting a Database response. If the timer expires and no response is received, the LSC shall route the call request to the local TDM PSTN trunk group (PRI or CAS) using the originally dialed commercial number.	5.3.2.23		Y	
180	On Commercial Cost Avoidance call requests that are re-routed to DSN numbers by the database, the LSC shall respond to MFSS or WAN SS AS-SIP signaling indicating that the call was rejected (i.e., an AS-SIP 4xx, 5xx, or 6xx response to an AS-SIP INVITE message), by overflowing these calls from the AS-SIP trunk group to the local TDM PSTN trunk group (PRI or CAS) using the originally dialed commercial number.	5.3.2.23		Y	
181	On Commercial Cost Avoidance call requests that are re-routed to DSN numbers by the database, the MFSS or WAN SS shall accept AS-SIP call requests from the LSC where the DSN number is identified as the called number. The MFSS or WAN SS shall also be capable of returning AS-SIP signaling to the calling LSC that indicates “404 Not Found,” “480 Temporarily Unavailable,” or “500 Server Internal Error.” The MFSS or WAN SS shall be capable of generating this AS-SIP signaling on its own, and shall be capable of relaying that AS-SIP signaling when it is received from a remote MFSS, remote WAN SS, or remote LSC.	5.3.2.23	Y		Y
182	For each RTS end user served by an LSC, the LSC shall be able to upload that user’s DSN phone number, PSTN phone number, and a unique LSC CCA-ID, Primary MFSS/WAN SS CCA-ID, and Backup MFSS/WAN SS CCA-ID to the RTS Routing Database.	5.3.2.23		Y	
183	The AS-SIP signaling appliance shall divert ALL unanswered RTS VoIP calls above the ROUTINE precedence level to a designated RTS DN for PCD (e.g., the number of an attendant console or group of attendant consoles).	5.3.2.25	Y	Y	C
184	Unanswered RTS VoIP calls above the ROUTINE precedence level shall not be forwarded to voicemail, and shall not be forwarded to ACD systems. Instead, they should divert to the PCD DN when the PCD time period expires.	5.3.2.25	Y	Y	C
185	Unanswered RTS VoIP calls at the ROUTINE precedence level shall still be forwarded to voicemail or to ACD systems (when Call Forwarding Don’t Answer is assigned to the called RTS DN), even though PCD is enabled and configured for the AS-SIP signaling appliance.	5.3.2.25	Y	Y	C

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
186	Calls above the ROUTINE precedence level that are destined to (directly dialed to) DNs assigned to voicemail or ACD systems shall only divert to the PCD DN as specified above (i.e., when they are unanswered at the voicemail or ACD system, and the PCD time period expires).	5.3.2.25	Y	Y	C
187	ROUTINE precedence level calls that are destined to (directly dialed to) DNs assigned to voicemail or ACD systems shall be allowed.	5.3.2.25	Y	Y	C
188	Incoming precedence calls to the attendant's listed DN, and incoming calls that are diverted to this attendant DN, shall be placed in a queue for the attendant console (or group of attendant consoles).	5.3.2.25	Y	Y	C
189	When a group of attendant consoles on the same LSC is used, and calls are either placed or diverted to the attendant console DN, call distribution across the Console Group shall be used to reduce excessive caller waiting times.	5.3.2.25	Y	Y	C
190	Incoming calls (placed and diverted) to the console DN shall be queued for attendant service by call precedence and time of arrival. The highest precedence call with the longest holding time in the queue shall be offered to an attendant first.	5.3.2.25	Y	Y	C
191	A recorded message of explanation (e.g., ATQA) shall be applied automatically to all the waiting calls in the Attendant Console queue (refer to Table 5.3.4-9, Announcements).	5.3.2.25	Y	Y	C
192	The RTS Attendant Console shall interoperate with PBAS/ASAC as described in • Section 5.3.2.7.2.1, PBAS/ASAC Requirements • Section 5.3.2.2.2.3, ASAC – Open Loop • Section 5.3.4.10, Precedence and Preemption The console shall be able to initiate all levels of RTS precedence calls (i.e., ROUTINE through FLASH-OVERRIDE).	5.3.2.26.1	Y	Y	C
193	The RTS Attendant Console shall provide a visual display of each precedence level and the calling number, for incoming direct dialed calls to the attendant, and diverted calls to the attendant (e.g., calls that reach the attendant through PCD).	5.3.2.26.2	Y	Y	C
194	The appliance and the attendant console shall give the attendant the ability to verify and override a busy line condition.	5.3.2.26.4	Y	Y	C
195	If the attendant uses BLV on a called line, and that called line (called EI) is busy, the appliance and the attendant console shall give an audible and visual "called line busy" indication back to the attendant.	5.3.2.26.4	Y	Y	C
196	The appliance and the attendant console shall prevent an attendant from activating BLV or Emergency Interrupt to called lines and called numbers that are located in the commercial network (the PSTN).	5.3.2.26.4	Y	Y	C
197	The appliance and the attendant console shall give the attendant the ability to use Emergency Interrupt to interrupt an existing call on a busy line, and inform the busy user of a new incoming call.	5.3.2.26.4	Y	Y	C
198	The appliance shall give selected destination EIs the ability to be exempt from Emergency Interrupt and attendant break-in.	5.3.2.26.4	Y	Y	C
199	The appliance and the attendant console shall have the ability to route all calls that are normally directed to the console to a separate night service deflection number. The night service deflection number shall be a fixed (preconfigured) or manually-selected DN.	5.3.2.26.5	Y	Y	C
200	When an attendant redirects an incoming call to a destination station, and that station is either busy or does not answer the call within a preset time, the appliance and the attendant console shall ensure that calling party on the redirected call is recalled automatically to the console. In this case, the appliance shall ensure that that the "recalled" call is returned to the console that originally processed the call.	5.3.2.26.6	Y	Y	C
201	The appliance and the attendant console shall have the ability to place calls (both directed to the attendant and diverted to the attendant) into a waiting queue.	5.3.2.26.7	Y	Y	C
202	The appliance and the attendant console shall ensure that calls in the attendant queue are not lost when a console is placed out of service or has its calls forwarded to a night service deflection number.	5.3.2.26.7	Y	Y	C
203	The LSCs that support SIP EIs MUST comply with the differentiated set of requirements defined for SIP EIs if they serve SIP EIs, and LSCs MUST comply with the differentiated set of requirements defined for H.323 and/or vendor-proprietary EIs if they serve H.323 and/or vendor-proprietary EIs.	5.3.4.7.1		Y	

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
204	All AS-SIP signaling appliances MUST comply with the SIP syntax and encoding rules set forth in RFC 3261. [RFC 3261, Section 25, Augmented BNF for the SIP Protocol]	5.3.4.7.1.1	Y	Y	Y
205	When an AS-SIP signaling appliance does not understand a header field in a request (and support for the header field is not a mandatory requirement under this specification), the AS-SIP signaling appliance MUST ignore that header field and continue processing the message. The AS-SIP signaling appliances MUST ignore any malformed header fields that are not necessary for processing requests.	5.3.4.7.1.3	Y	Y	Y
206	When an AS-SIP signaling appliance, that is implemented as a SIP proxy, receives a SIP Request message, 2xx response, or 18x response, then the AS-SIP signaling appliance MUST add a Record-Route header whereby the userinfo part of the SIP URI is a unique identifier for the AS-SIP signaling appliance and an IP address is used for the host name.	5.3.4.7.1.3c	Y	Y	Y
207	All AS-SIP signaling appliances MUST be call stateful.	5.3.4.7.1.4	Y	Y	Y
208	Upon receipt of a new request, AS-SIP signaling appliances MUST perform request validation, route information preprocessing, determine request targets, perform request forwarding, perform response processing, process timer C, handle transport error, handle CANCEL processing, and perform proxy route processing according to RFC 3261	5.3.4.7.1.5	Y	Y	Y
209	All AS-SIP signaling appliances MUST support generation of the long form of the SIP header fields along with the receipt and processing of the long form of the SIP header fields.	5.3.4.7.1.7	Y	Y	Y
210	All AS-SIP signaling appliances MUST support receiving and processing the compact form of the SIP header fields.	5.3.4.7.1.8	Y	Y	Y
211	All AS-SIP signaling appliances serving IP EIs MUST support the offer/answer model for the Session Description Protocol (SDP).	5.3.4.7.1.9	Y	Y	Y
212	If an LSC receives a call request from a served IP EI and the LSC has been unable to establish a TLS connection with its EBC and is unable to do so upon receipt of the INVITE, then the AS-SIP signaling appliance MUST ensure that the IP EI plays the Isolated Code Announcement (ICA) and terminates the call request and MUST send an alarm to the NMS.	5.3.4.7.1.10		Y	
213	When an SS receives an INVITE from either a served LSC or another SS where the Request-URI has a DSN telephone number for which the SS has no entry in its Location Server, then the SS MUST respond with a 404 (Not Found) response code.	5.3.4.7.1.12	Y		Y
214	When an LSC receives an inbound INVITE from its primary (or secondary) SS whose Request-URI has a DSN telephone number for which the LSC has no entry in its Location Server, then the LSC MUST respond with a 404 (Not Found) response message.	5.3.4.7.1.13		Y	
215	The LSCs serving IP EIs MUST ensure that all outbound INVITEs forwarded onto the UC WAN include a Supported header with the option tag “100rel.”	5.3.4.7.1.14		Y	
216	When an AS-SIP signaling appliance receives an INVITE (having an sdp offer) and will be interworking the signaling to the TDM network, then the AS-SIP signaling appliance MUST return an sdp answer in the first non-failure reliable provisional response.	5.3.4.7.1.15	Y	Y	Y
217	When an LSC receives an INVITE (having an sdp offer) intended for a served IP EI, then the AS-SIP signaling appliance MUST NOT return an sdp answer in any provisional response and MUST only place the sdp answer in the 200 response.	5.3.4.7.1.16		Y	
218	When an AS-SIP signaling appliance receives an Empty INVITE (i.e., an INVITE that does not include an sdp offer) and said AS-SIP signaling appliance will be interworking the signaling to the TDM network, then the AS-SIP signaling appliance MUST send an sdp offer in the first reliable non-failure provisional response (1xx response code greater than a 100 response code).	5.3.4.7.1.17	Y	Y	Y
219	When an AS-SIP signaling appliance receives an Empty INVITE intended for a served IP EI, then the AS-SIP signaling appliance MUST NOT send an sdp offer in any provisional response (1xx response code greater than a 100 response code) and MUST only send the sdp offer in the 200 response.	5.3.4.7.1.18	Y	Y	Y
220	When an AS-SIP signaling appliance that is interworking SIP signaling with the TDM network receives a 180 (Ringing) response from the IP network, the AS-SIP signaling appliance MUST ensure that the appropriate ring back tone (e.g., ring back, precedence ring back) is generated on the TDM network.	5.3.4.7.1.19	Y	Y	Y

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
221	Announcements are not sent in-band on the DSN TDM network; therefore, when an AS-SIP signaling appliance that is interworking SIP signaling with the TDM network receives a 480 (Temporarily Unavailable), 486 (Busy Here), or 488 (Not Acceptable Here) response from the IP network with either no Reason header or a Reason header that does NOT have a preemption cause, the AS-SIP signaling appliance does NOT generate an announcement to be sent to the TDM network, rather it sends either a REL with Q.850 cause code 46 precedence call blocked (in the case of SS7 ISUP) or a Disconnect (in the case of ISDN) with the appropriate cause code message to the TDM network.	5.3.4.7.1.20	Y	Y	Y
222	An LSC that receives an outbound call request from a served IP EI MAY include an audio media feature tag and a video media feature tag, as appropriate, in the Contact header field of the INVITE message.	5.3.4.7.1.21		Y	
223	The AS-SIP signaling appliances are NOT required to process and act on the audio media tag and the video media tag in the Contact header but all intermediary AS-SIP signaling appliances MUST preserve the audio media tag (if present) and the video tag (if present) when forwarding the INVITE. (i.e., intermediary AS-SIP signaling appliances MUST NOT strip off or modify the media feature tags).	5.3.4.7.1.22	Y	Y	Y
224	When an LSC receives a call request from a served IP EI intended for a destination outside the enclave, then the AS-SIP signaling appliance MUST generate the P-Asserted-Identity header.	5.3.4.7.1.23		Y	
225	The LSC serving the AS-SIP EI MUST support authentication of the AS-SIP EIs. The user of the AS-SIP EI is required to perform user authentication to the LSC when initiating precedence call requests.	5.3.4.7.1.26		Y	
226	The LSCs serving IP EIs (other than AS-SIP EIs) MUST support authentication of the IP EIs. The user of the IP EI is required to perform user authentication to the LSC when initiating precedence call requests.	5.3.4.7.1.31		Y	
227	When an LSC serving H.323 and/or proprietary EIs receives a request that contains a Require header field with one or more option tags that it does not understand, then it MUST return a 420 (Bad Extension) response code. The response MUST include an Unsupported header field listing those option tags the element did not understand.	5.3.4.7.1.35		Y	
228	The LSCs and AS-SIP EIs MUST support the generating, receiving, and processing of SIP CANCEL requests.	5.3.4.7.2.2		Y	
229	The AS-SIP signaling appliances MUST support the keep-alive mechanism for SIP sessions.	5.3.4.8.1.1	Y	Y	Y
230	The AS-SIP signaling appliances MUST support the generating, receiving, and processing of the Session-Expires and Min-SE header fields.	5.3.4.8.1.3	Y	Y	Y
231	The AS-SIP signaling appliances MUST support the 422 (Session Interval Too Small) response code.	5.3.4.8.1.4	Y	Y	Y
232	The AS-SIP signaling appliances MUST support the option tag “timer” for use with the Supported and Require header fields; however, an AS-SIP signaling appliance acting as a UAC or a SIP EI acting as a UAC MUST NOT place the option tag “timer” in either a Require header or a Proxy-Require header.	5.3.4.8.1.4	Y	Y	Y
233	When an AS-SIP signaling appliance receives an outbound request from a served H.323 or proprietary IP EI, then the AS-SIP signaling appliance MUST operate in accordance with the UAC behavior (when responsible for performing the refresh).	5.3.4.8.1.8	Y	Y	Y
234	When an AS-SIP signaling appliance receives a call request from another AS-SIP signaling appliance, and the destination is a served H.323 or proprietary IP EI, then the AS-SIP signaling appliance MUST operate in accordance with the UAS behavior (when responsible for performing the refresh).	5.3.4.8.1.10	Y	Y	Y
235	When SDP information is present in a SIP message, the SIP message MUST have a content-type header having the MIME Content-Type “application/sdp”.	5.3.4.9.1.2	Y	Y	Y
236	The SDP parser in the AS-SIP signaling appliance (and all AS-SIP EIs, including AS-SIP video conferencing EIs) MUST be able to accept and handle without error any of the SDP line types enumerated in RFC 2327 even if the application ignores the contents.	5.3.4.9.1.3	Y	Y	Y
237	The precedence level of the call request MUST be set forth in a SIP Resource-Priority header field whose syntax is in accordance with RFC 4412, as modified in UCR 2008, Section 5.3.4.10.2	5.3.4.10.2.1	Y	Y	C
238	Video telephony EIs MUST, as the default configuration, require an end user wishing to place a call that includes video, to affirmatively signal the intention to include video to the EI every time the caller wishes to engage in a video telephony call.	5.3.4.12.1.1	Y	Y	C

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
239	Every time a caller requests a video telephony call and the video portion of the telephony call is successfully established, then the video telephony EI MUST provide the user with an affirmative confirmation that the video is enabled either before, or upon successful completion of, session establishment.	5.3.4.12.1.2	Y	Y	C
240	When an INVITE with an sdp offer that includes both audio and video capabilities is received by an LSC serving a destination EI that supports video telephony, then when the call request is received by the destination EI the destination EI MUST indicate to the callee that a telephony call requesting video connectivity has been received.	5.3.4.12.2.1	Y	Y	C
241	Every time a callee accepts a video telephony call and the video portion of the telephony call is successfully established, then the video telephony EI MUST provide the user with an affirmative confirmation that the video is enabled either before or upon successful session establishment.	5.3.4.12.2.3	Y	Y	C
242	AS-SIP Signaling appliances must follow call flows depicted in section 5.3.4.13 for all call features and calling services.	5.3.4.13	Y	Y	Y
243	AS-SIP Signaling appliances must follow requirements depicted in section 5.3.4.14 for all IP to TDM and TDM to IP translations.	5.3.4.14	Y	Y	Y
244	When an interworking AS-SIP signaling appliance receives a request that contains a Require header field with one or more option-tags that it does not understand, then the interworking AS-SIP signaling appliance MUST return a 420 (Bad Extension) response. The response MUST include an Unsupported header field listing those option-tags the element did not understand.	5.3.4.16.1.1	Y	Y	Y
245	All outbound INVITEs generated by an interworking AS-SIP signaling appliance MUST include a Supported header with the option tag "100rel."	5.3.4.16.1.2	Y	Y	Y
246	Interworking AS-SIP signaling appliances MUST support generating and receiving SIP INVITE requests. Interworking AS-SIP signaling appliances MUST support generating and receiving SIP re-INVITES.	5.3.4.16.2.1	Y	Y	Y
247	Interworking AS-SIP signaling appliances MUST support generating and receiving SIP CANCEL requests.	5.3.4.16.2.2	Y	Y	Y
248	Interworking AS-SIP signaling appliances MUST support generating and receiving SIP OPTIONS requests.	5.3.4.16.2.4	Y	Y	Y
249	Interworking AS-SIP signaling appliances MUST support generating and receiving SIP BYE requests	5.3.4.16.2.5	Y	Y	Y
250	Interworking AS-SIP signaling appliances MUST support generating and receiving SIP ACK requests	5.3.4.16.2.6	Y	Y	Y
251	Interworking AS-SIP signaling appliances MUST support generating and receiving the SIP PRACK method. Interworking AS-SIP signaling appliances MUST support use of the option tag "100rel" with the Require header and Supported header, and MUST support the use of header fields RAck and RSeq.	5.3.4.16.2.8	Y	Y	Y
252	Interworking AS-SIP signaling appliances MUST support generating and receiving the SIP UPDATE method	5.3.4.16.2.9	Y	Y	Y
253	Inter-working AS-SIP signaling appliances MUST be capable of receiving/processing REFER requests, the Refer-To header, and the REFER event package.	5.3.4.16.2.10	Y	Y	Y
254	Interworking AS-SIP signaling appliances MUST support the NOTIFY method for event notification.	5.3.4.16.2.12	Y	Y	Y
255	Interworking AS-SIP signaling appliances MUST, in adherence with the enumerated RFCs, be capable of generating, receiving, and processing the SIP headers listed in UCR 2008 Section 5.3.4.16.3.1	5.3.4.16.3.1	Y	Y	Y
256	The From header MUST include a tag field as specified in RFC 3261, Section 19.3.	5.3.4.16.3.3	Y	Y	Y
257	The To header of a request that is part of a dialog MUST include a tag field as specified in RFC 3261, Section 19.3.	5.3.4.16.3.4	Y	Y	Y
258	Interworking AS-SIP signaling appliances MUST support the use of option tags for the Require, Supported, and Unsupported headers.	5.3.4.16.3.5	Y	Y	Y
259	When the interworking LSC sends an initial AS-SIP INVITE to its local EBC intended for its SS, the interworking LSC MUST add two Route header field values, which either takes the form of a route set comprising two Route headers where the first Route header is the sip uri for the EBC at the enclave and the second Route header is the sip uri for the EBC serving the SS, or takes the form of one Route header with two comma-separated field values.	5.3.4.16.3.6		Y	

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
260	When an interworking SS forwards an initial AS-SIP INVITE to a peer SS, then the interworking SS MUST add a route set comprising two Route headers where the first Route header is the SIP URI for the EBC that serves the interworking SS, and the second Route header is the SIP URI for the EBC serving the peer SS.	5.3.4.16.3.7	Y		Y
261	When an interworking AS-SIP signaling appliance generates an outbound AS-SIP request, the interworking AS-SIP signaling appliance MUST add its own VIA header to the AS-SIP request.	5.3.4.16.3.8	Y	Y	Y
262	When an interworking AS-SIP signaling appliance receives a SIP response to be translated into TDM signaling, then the interworking AS-SIP signaling appliance operates as the UAC for SIP purposes.	5.3.4.16.3.9	Y	Y	Y
263	When an interworking AS-SIP signaling appliance receives an inbound SIP request to be translated into TDM signaling, then the AS-SIP signaling appliance operates as the UAS for SIP purposes.	5.3.4.16.3.10	Y	Y	Y
264	When an interworking AS-SIP signaling appliance generates a SIP response on behalf of a signaling message received from the TDM network, then before forwarding the SIP response the interworking AS-SIP signaling appliance MUST include the VIA headers received in the corresponding SIP request.	5.3.4.16.3.11	Y	Y	Y
265	When an interworking AS-SIP signaling appliance operating as an originating gateway receives an IAM from the TDM network and sends an INVITE to another AS-SIP signaling appliance (SS or LSC), then the interworking AS-SIP signaling appliance MUST add a CCA-ID parameter to the SIP URI of the Contact header populated with its unique identifier before forwarding the INVITE onward to the next AS-SIP signaling appliance.	5.3.4.16.3.12	Y	Y	Y
266	Interworking AS-SIP signaling appliances MUST support generating, receiving, and processing the provisional (1xx) response codes: 100 (Trying), 180 (Ringing), and 183 (Session Progress).	5.3.4.16.4.1	Y	Y	Y
267	Interworking AS-SIP signaling appliances MUST support generating and receiving/processing the successful (2xx) response codes: 200 (OK) [RFC 3261, Section 21.2, 200 OK] and 202 (Accepted)	5.3.4.16.4.2	Y	Y	Y
268	Interworking AS-SIP signaling appliances MUST support generating and receiving/processing the request failure (4xx) response codes: 400 (Bad Request), 401 (Unauthorized), 403 (Forbidden), 404 (Not Found), 405 (Method Not Allowed), 406 (Not Acceptable), 407 (Proxy Authentication Required), 408 (Request Timeout), 410 (Gone), 413 (Request Entity Too Large), 414 (Request-URI Too Long), 415 (Unsupported Media Type), 416 (Unsupported URI Scheme), 417 (Unknown Resource-Priority), 420 (Bad Extension), 421 (Extension Required), 422 (Session Interval Too Small), 423 (Interval Too Brief), 480 (Temporarily Unavailable), 481 (Call/Transaction Does Not Exist), 482 (Loop Detected), 483 (Too Many Hops), 484 (Address Incomplete), 485 (Ambiguous), 486 (Busy Here), 487 (Request Terminated), 488 (Not Acceptable Here), and 491 (Request Pending).	5.3.4.16.4.4	Y	Y	Y
269	Interworking AS-SIP signaling appliances upon properly receiving a CANCEL request for an INVITE MUST first send a 200 (OK) response code to the CANCEL, and then follow up with a 487 (Request Terminated) response code to the INVITE.	5.3.4.16.4.5	Y	Y	Y
270	Interworking AS-SIP signaling appliances MUST support generating and receiving/processing the server failure (5xx) response codes: 500 (Server Internal Error), 501 (Not Implemented), 502 (Bad Gateway), 503 (Service Unavailable), 504 (Server Timeout), 505 (Version Not Supported), 513 (Message Too Large) [RFC 3261, Section 21.5, Server Failure 5xx], and 580 (Precondition Failure)	5.3.4.16.4.6	Y	Y	Y
271	Interworking AS-SIP signaling appliances MUST support generating and receiving/processing the global failures (6xx) response codes: 600 (Busy Everywhere), 603 (Decline), 604 (Does Not Exist Anywhere), and 606 (Not Acceptable).	5.3.4.16.4.7	Y	Y	Y
272	When an interworking AS-SIP signaling appliance receives an outbound request from the PSTN (i.e., the interworking AS-SIP signaling appliance is operating as an originating gateway) and the destination is NOT an IP EI directly served by the interworking AS-SIP signaling appliance, then the interworking AS-SIP signaling appliance MUST operate in accordance with the UAC behavior set forth in RFC 4028.	5.3.4.17.1.1	Y	Y	Y
273	When an interworking AS-SIP signaling appliance acting as a terminating gateway receives a call request from another AS-SIP signaling appliance, then the interworking AS-SIP signaling appliance MUST operate in accordance with the UAS behavior set forth in RFC 4028.	5.3.4.17.1.3	Y	Y	Y

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
274	Interworking AS-SIP signaling appliances MUST support the option tag “resource-priority” for use with the Require header.	5.3.4.18.3.1	Y	Y	Y
275	The interworking AS-SIP signaling appliance MUST receive and accept a Require header field with the option tag “resource-priority” in the INVITE, UPDATE, and REFER messages. Interworking AS-SIP signaling appliances MUST NOT reject the message with a 420 (Bad Extension) response code, but rather it MUST accept the request and translate it into the appropriate TDM signaling message as required.	5.3.4.18.3.2	Y	Y	Y
276	If an interworking AS-SIP signaling appliance receives an inbound ROUTINE call request over the IP network for a destination on the TDM network and the interworking AS-SIP signaling appliance has insufficient bandwidth-related resources (e.g., lack of circuit-switched trunk capacity for bearer traffic) to handle the call request, the interworking AS-SIP signaling appliance MUST reply with a 488 (Not Acceptable Here) response code and SHOULD include a Warning header with warning code 370 (Insufficient Bandwidth).	5.3.4.18.4.5	Y	Y	Y
277	If an interworking AS-SIP signaling appliance receives an inbound precedence call request (i.e., with precedence level PRIORITY or above) over the IP network for a destination on the TDM network and the interworking AS-SIP signaling appliance has insufficient bandwidth-related resources (e.g., lack of circuit-switched trunk capacity for bearer traffic) to handle the call request, and if there are insufficient existing calls (and/or call requests) of lower precedence whose removal would provide the necessary resources to support the pending call request, then: - The interworking AS-SIP signaling appliance MUST reply with a 488 (Not Acceptable Here) response code and SHOULD include a Warning header with warning code 370 (Insufficient Bandwidth), and - The AS-SIP signaling appliance serving the calling IP EI MUST arrange for a BPA to be played to the calling IP EI before terminating the call.	5.3.4.18.4.6	Y	Y	Y
278	When an interworking AS-SIP signaling appliance receives a precedence call request from the IP network that it translates and forwards onto the TDM network and the response from the TDM network is a REL with Q.850 cause code 46 precedence call blocked (in the case of SS7 ISUP) or a Disconnect with Q.850 cause code 46 precedence call blocked (in the case of ISDN), then the interworking AS-SIP signaling appliance MUST generate a 488 (Not Acceptable Here) response that SHOULD include a “Warning” header with warning code 370 (Insufficient Bandwidth) with no Reason header that it sends onto the IP network.	5.3.4.18.6.2	Y	Y	Y
279	Interworking AS-SIP signaling appliances directly serving IP EIs MUST comply with the requirements in UCR 2008, Section 5.3.4.10.3.3.2, Implementing the Network Preemption. In addition, interworking AS-SIP signaling appliances directly serving IP EIs MUST meet the enumerated requirements in section 5.3.4.18.6.3.2.	5.3.4.18.6.3	Y	Y	Y
280	AS-SIP signaling appliances must follow all call flows depicted in UCR 2008 Section 5.3.4.19 for all supplementary services.	5.3.4.19	Y	Y	Y
281	The product shall support dual IPv4 and IPv6 stacks as described in RFC 4213.	5.3.5.4	Y		Y
282	Dual stack end points or Call Control Agents shall be configured to choose IPv4 over IPv6.	5.3.5.4	Y		Y
283	All nodes that are “IPv6-capable” shall be carefully configured and verified that the IPv6 stack is disabled until it is deliberately enabled as part of a risk management strategy.	5.3.5.4	Y		Y
284	The product shall support the IPv6 format as described in RFC 2460 and updated by RFC 5095.	5.3.5.4	Y		Y
285	The product shall support the transmission of IPv6 packets over Ethernet networks using the frame format defined in RFC 2464.	5.3.5.4	Y		Y
286	The product shall support a minimum MTU of 1280 bytes.	5.3.5.4.1	Y		Y
287	The product shall not use the Flow Label field as described in RFC 2460. The product shall be capable of setting the Flow Label field to zero when originating a packet. The product shall not modify the Flow Label field when forwarding packets. The product shall be capable of ignoring the Flow Label field when receiving packets.	5.3.5.4.2	Y		Y
288	The product shall support the IPv6 Addressing Architecture as described in RFC 4291.	5.3.5.4.3	Y		Y
289	The product shall support the IPv6 Scoped Address Architecture as described in RFC 4007.	5.3.5.4.3	Y		Y

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
290	The product shall support Neighbor Discovery for IPv6 as described in RFC 2461 and RFC 4861.	5.3.5.4.5	Y		Y
291	The product shall not set the override flag bit in the Neighbor Advertisement message for solicited advertisements for anycast addresses or solicited proxy advertisements.	5.3.5.4.5	Y		Y
292	When a valid “Neighbor Advertisement” message is received by the product and the product neighbor cache does not contain the target’s entry, the advertisement shall be silently discarded.	5.3.5.4.5	Y		Y
293	When a valid “Neighbor Advertisement” message is received by the product and the product neighbor cache entry is in the INCOMPLETE state when the advertisement is received and the link layer has addresses and no target link-layer option is included, the product shall silently discard the received advertisement.	5.3.5.4.5	Y		Y
294	When address resolution fails on a neighboring address, the entry shall be deleted from the product’s neighbor cache.	5.3.5.4.5	Y		Y
295	The product shall support the ability to configure the product to ignore Redirect messages. The product shall only accept Redirect messages from the same router as is currently being used for that destination.	5.3.5.4.5.1	Y		Y
296	If the product supports routing functions, the product shall inspect valid router advertisements sent by other routers and verify that the routers are advertising consistent information on a link and shall log any inconsistent router advertisements. The product shall prefer routers that are reachable over routers whose reachability is suspect or unknown.	5.3.5.4.5.2	Y		Y
297	The product shall support manual assignment of IPv6 addresses.	5.3.5.4.6	Y		Y
298	The product shall support the ICMPv6 as described in RFC 4443. The product shall have a configurable rate limiting parameter for rate limiting the forwarding of ICMP messages.	5.3.5.4.7	Y		Y
299	The product shall support the capability to enable or disable the ability of the product to generate a Destination Unreachable message in response to a packet that cannot be delivered to its destination for reasons other than congestion.	5.3.5.4.7	Y		Y
300	The product shall support the enabling or disabling of the ability to send an Echo Reply message in response to an Echo Request message sent to an IPv6 multicast or anycast address.	5.3.5.4.7	Y		Y
301	The product shall validate ICMPv6 messages, using the information contained in the payload, before acting on them.	5.3.5.4.7	Y		Y
302	The product shall support MLD as described in RFC 2710.	5.3.5.4.8	Y		Y
303	For traffic engineering purposes, the bandwidth required per voice subscriber is calculated to be 110.0 kbps (each direction) for each IPv6 call.	5.3.5.4.11	Y		Y
304	The product shall forward packets using the same IP Version as the Version in the received packet.	5.3.5.4.12	Y		Y
305	The product shall use the Alternative Network Address Types (ANAT) semantics for the Session Description Protocol (SDP) in accordance with RFC 4091 when establishing media streams from dual-stacked appliances for AS-SIP signaled sessions.	5.3.5.4.12	Y		Y
306	The product shall prefer any IPv4 address to any IPv6 address when using ANAT semantics.	5.3.5.4.12	Y		Y
307	The product shall place the option tag “SDP-ANAT” in a Required header field when using ANAT semantics in accordance with RFC 4092.	5.3.5.4.12	Y		Y
308	The products shall support Differentiated Services as described in RFC 2474 for a voice and video stream in accordance with Section 5.3.2, Assured Services Requirements, and Section 5.3.3, Network Infrastructure E2E Performance Requirements, plain text DSCP plan.	5.3.5.4.14	Y		Y
309	The LSC must meet all requirements for FCAPS Management and audit logs as listed in UCR 2008 section 5.3.2.7.2.6	5.3.2.7.2.6		Y	
310	The physical interface between the DISA VVoIP EMS and the network components (i.e., LSC, MFSS, EBC, CE Router) is a 10/100-Mbps Ethernet interface. The interface will work in either of the two following modes using auto-negotiation: IEEE, Ethernet Standard 802.3, 1993; or IEEE, Fast Ethernet Standard 802.3u, 1995.	5.3.2.4.4	Y	Y	Y
311	Redundant physical Ethernet interfaces are required for signaling and bearer traffic. If the primary signaling and bearer Ethernet interface fails, then traffic shall be switched to the backup signaling and bearer Ethernet interface.	5.3.2.4.4	Y	Y	Y

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
312	The MFSS shall provide a single, common interface to the DISA NMS. The single interface shall provide access to MFSS features and functions for both the TDM and SS side of the MFSS.	5.3.2.8.3.1	Y		
313	The MFSS-to-NMS interface shall be an Ethernet connection as specified in Section 5.3.2.4.4, VVoIP NMS Interface Requirements.	5.3.2.8.3.1	Y		
314	As specified in Section 5.3.2.4.4, VoIP NMS Interface Requirements, the MFSS, WAN SS, and LSC components shall support at least one pair of physical Ethernet management interfaces at the component level (not at the device level). One of these Ethernet management interfaces shall be used for component-level communication with a Local EMS. The other Ethernet management interface shall be used for component-level communication with the remote VVoIP EMS.	5.3.2.17.2	Y	Y	Y
315	A network appliance shall have Operations interfaces that provide a standard means by which management systems can directly or indirectly communicate with and, thus, manage the various network appliances in the DISN.	5.3.2.17.2	Y	Y	Y
316	There shall be a local craftsperson interface (Craft Input Device (CID)) for OA&M for all VVoIP network components.	5.3.2.17.2	Y	Y	Y
317	The network appliances shall provide NM data to the external VVoIP EMS.	5.3.2.17.2	Y	Y	Y
318	A network appliance shall communicate with an external Voice and Video management system by a well-defined, standards-based management interface using an industry-accepted management protocol.	5.3.2.17.2	Y	Y	Y
319	Communications between VVoIP EMS and the VVoIP network appliances shall be via IP.	5.3.2.17.2	Y	Y	Y
320	A network appliance shall issue state change notifications for changes in the states of replaceable components, including changes in operational state or service status, and detection of new components.	5.3.2.17.2	Y	Y	Y
321	A network appliance shall be provisioned by the VVoIP EMS with the address, software, and OSI Layer 4 port information associated with its Core Network interfaces.	5.3.2.17.2	Y	Y	Y
322	A network appliance shall be capable of maintaining and responding to VVoIP EMS requests for resource inventory, configuration, and status information concerning Core Network interface resources (e.g., IP or MAC addresses) that have been installed and placed into service.	5.3.2.17.2	Y	Y	Y
323	Network appliances that provide voice and video call service shall have the capability to invoke traffic flow (NM) controls as detailed in Section 5.3.2.18, Network Management Requirements of Appliance Functions.	5.3.2.17.2	Y	Y	Y
324	A network appliance shall be capable of setting the Administrative state and maintaining the Operational state of each Core Network interface, and maintaining the time of the last state change.	5.3.2.17.2	Y	Y	Y
325	Alarm messages must be distinguishable from administrative log messages.	5.3.2.17.3.1.1	Y	Y	Y
326	The NEs shall detect their own fault (alarm) conditions.	5.3.2.17.3.1.2	Y	Y	Y
327	The NEs shall generate alarm notifications.	5.3.2.17.3.1.3	Y	Y	Y
328	The network elements shall send the alarm messages in NRT. More than 99.95 percent of alarms shall be detected and reported in NRT. Near Real Time is defined as event detection and alarm reporting within 5 seconds of the event, excluding transport time.	5.3.2.17.3.1.4	Y	Y	Y
329	The network components shall send alarm messages in SNMPv3 format.	5.3.2.17.3.1.5	Y	Y	Y
330	Capability to access and modify configuration data by the VVoIP EMS shall be controllable by using an access privileges function within the network appliance.	5.3.2.17.3.2.1	Y	Y	Y
331	The VVoIP NEs shall be able to receive and respond to remote NM commands.	5.3.2.17.3.4.2	Y	Y	Y
332	When ASAC budgets are reduced, by NM action, below the current budget allocation, any previous sessions (regardless of precedence level) in excess of the new budget shall be allowed to terminate naturally. This assumes that the CE Router queue bandwidths would not be reduced until the LSC session count fell below or equal to the newly commanded reduced budget, to prevent the corruption of existing sessions.	5.3.2.17.3.4.2.2	Y	Y	Y
333	The LSC, MFSS, and WAN SS shall have the capability of setting the percentage of calls to be blocked to the designated destination(s).	5.3.2.17.3.4.2.7	Y	Y	Y
334	FLASH and FLASH-OVERRIDE calls shall not be affected by NM controls.	5.3.2.17.3.4.2.7	Y	Y	Y

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
335	Within IP, directionalization is controlled by designating all or part of the call budget as inbound (i.e., local destination) and/or outbound (i.e., local origination). The default is no designation (i.e., calls up to the total budget can be inbound or outbound in any combination). It does not change the total budget, only the sourcing direction of the budget; therefore, there is no impact to the router queue bandwidths.	5.3.2.17.3.4.2.10	Y	Y	Y
336	Within IP, the routing of all traffic (i.e., VVoIP and non-VVoIP) is handled via MPLS in the DISN core. The MPLS automatically finds the most effective route for the traffic.	5.3.2.17.3.4.2.11	Y	Y	Y
337	The WAN-level ASAC must be able to account for each subtended LSC under its control. Therefore, the MFSS and WAN SS ASAC must be able to set call budgets for multiple LSC locations via the VVoIP EMS and local EMS access points.	5.3.2.17.3.4.2.13	Y		Y
338	The LSC-level ASAC is required to only account for itself. Therefore, the LSC ASAC must be able to set call budgets for only the PEI/AEIs under its control via the VVoIP EMS and local EMS access points.	5.3.2.17.3.4.2.13		Y	
339	The product shall have the capability of setting a PEI/AEI's maximum allowed precedence level for originating a call. This is a "subscriber class mark feature," which is controlled by the LSC system administrator.	5.3.2.17.3.4.2.14		Y	
340	The product shall have the capability of controlling the destination(s) that a PEI or AEI is restricted from calling. This is a subscriber class mark feature that is controlled by the LSC system administrator.	5.3.2.17.3.4.2.14		Y	
341	The ASAC must provide the separate counts for voice and video, in 5-minute intervals. The MFSS and WAN SS ASAC must provide these counts for each of the subtended LSCs under its control, while the LSC is only to provide these counts for the PEIs/AEIs that it controls.	5.3.2.18.2	Y	Y	Y
342	A switching network appliance shall acquire, activate, and manage a CCA software download as directed by the Local EMS. The CCA software may be managed on a per CCA hardware component basis.	5.3.2.18.3.1.1	Y	Y	Y
343	The CCA shall be able to manage the following parameters in the CCA from the VVoIP EMS: • CCA Identification parameter • Recording Office Identification parameter	5.3.2.18.3.1.1	Y	Y	Y
344	The CCA shall manage the activation and deactivation of service features. The CCA shall maintain data for the media server and UFS functions it interacts with. The CCA shall be able to create a backup and manage restoration of configuration data by placing its stable data and changes to the latest configuration in a nonvolatile storage device.	5.3.2.18.3.2	Y	Y	Y
345	A CCA shall meet all applicable Operations Technology Generic Requirements (OTGR) for switching system NE trouble isolation in Telcordia Technologies GR-474-CORE. A CCA shall perform root-cause analysis for any faults within its purview, report the root cause, and suppress the reporting of non-root-cause conditions. A CCA shall support the ability to perform internal diagnostics on its call processing functionality and internal resources, initiated either locally or upon request by the VVoIP EMS.	5.3.2.18.3.3	Y	Y	Y
346	The CCA shall provide trunk group-related traffic measurements as specified in Telcordia Technologies GR-477-CORE, Section 4.1.3. For all calls originating at a CCA, the CCA shall monitor call set-up delay statistics, including delay incurred as part of the set-up of the core network bearer connection.	5.3.2.18.3.5	Y	Y	Y
347	An MG shall manage logical and physical resource inventory information. An MG shall issue an autonomous notification to the VVoIP EMS whenever a new inventory or capabilities are added, or configuration is changed through local management activity. An MG maintains the information related to service features and data, including the management of service logic.	5.3.2.18.5.1	Y	Y	Y
348	An MG shall manage current MG state and status information about its installed major components, line and plug-in cards, and processes.	5.3.2.18.5.1.2	Y	Y	Y
349	Upon the detection or clearing of alarm conditions, the MG shall generate and forward, based on filtering criteria, a notification to the VVoIP EMS. An MG shall support queries for alarm status, state, and current problem information. An MG shall monitor, detect, and generate alarm conditions and states associated with hardware, functional components, system interfaces, and logical resources (e.g., trunk terminations, tone and announcement generators, media content detectors, signal processors, echo control devices).	5.3.2.18.5.2.1	Y	Y	Y
350	An MG shall perform root-cause analysis for any faults within its purview, report the root cause, and suppress the reporting of non-root-cause conditions.	5.3.2.18.5.2.2	Y	Y	Y

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
351	An MG shall, on request or per a pre-established schedule, run diagnostics on internal resources, hardware, or software, and report the result to the VVoIP EMS.	5.3.2.18.5.2.3	Y	Y	Y
352	An MG shall provide both local and remote loopback capabilities for the digital interfaces that terminate at the MG ports.	5.3.2.18.5.2.3	Y	Y	Y
353	Upon receiving a request from the VVoIP EMS or by an established schedule, an MG shall provide a report of a parameter's present or history counters.	5.3.2.18.5.3.1	Y	Y	Y
354	An MG shall generate TCAs to notify the VVoIP EMS when a thresholded count exceeds its threshold during a measurement interval.	5.3.2.18.5.3.2	Y	Y	Y
355	The MG shall manage interexchange trunk (between MG and SSP), trunk group, trunk, and physical resource inventory and configuration data. The MG shall manage MG termination-related status information.	5.3.2.18.5.4	Y	Y	Y
356	An MG shall, on request or on schedule, run diagnostics on internal resources and hardware, run checks on software, and report the results to the VVoIP EMS. The MG shall provide test access to external test equipment for passively monitoring the traffic through the MG interfaces. This passive monitoring shall not degrade the performance of traffic.	5.3.2.18.5.5	Y	Y	Y
357	The MG shall receive voice-grade analog line configuration data from the VVoIP EMS upon service activation.	5.3.2.18.5.7	Y	Y	Y
358	The MG shall provide diagnostic tests to detect and verify faults, such as low loop resistance or ground conditions, or any other faults within the MG that could cause false ring trip or false answer.	5.3.2.18.5.8	Y	Y	Y
359	The MG shall support the collection of the standard DS1, DS3, Physical Layer Convergence Protocol (PLCP), SONET, and ISDN BRI line performance monitoring requirements, as defined in Telcordia Technologies GR-820-CORE, for applicable interfaces.	5.3.2.18.5.9	Y	Y	Y
360	For the selected recording format that is chosen, of all the call information that will be provided, the following call data shall be provided in the record data: 1. Host Name of the CCA controlling the call processing. 2. Start Date of call (In Julian or Calendar). 3. Start Time of Call (Hour + Minute + Second). 4. Elapsed Time of Call and/or Stop Time of call. 5. Calling Number. 6. Called Number (included all dialed digits).	5.3.2.19.2.1	Y	Y	Y
361	For the selected recording format that is chosen, of all the call information that will be provided, the following call data shall be provided in the record data if it applies to the call: Conference Call Indicator.	5.3.2.19.2.1	Y	Y	Y
362	The product shall provide a voice quality record at the completion of each voice session. The voice quality record shall be included in the CDR that the LSC, MFSS, or WAN SS generates for that session, and shall conform to the E-Model, as described in TIA TSB-116-A, and ITU-T Recommendation G.107. The voice quality record shall contain the calculated R-Factor for the Voice session per TIA TSB-116-A.	5.3.2.19.2.1.1	Y	Y	Y
363	As part of the voice quality record, the product shall provide the raw voice session statistics that are used to make the R-Factor calculation to include, as a minimum, the latency, packet loss, Equipment Impairment Factor (Ie), and the TCLw. The product shall provide the jitter for the session.	5.3.2.19.2.1.1	Y	Y	Y
364	The product shall generate an alarm to the VVoIP EMS when the session R-Factor calculation in the CDR fails to meet a configurable threshold. By default, the threshold shall be an R-Factor value of 80, which is equivalent to an MOS value of 4.0.	5.3.2.19.2.1.1	Y	Y	Y
365	The mass storage in the BA must be non-volatile. The mass storage in the BA must be able to retain at least five average-busy-season business days of AMA data. (NOTE: This is needed to provide adequate capacity for high-volume storage of CDRs.)	5.3.2.19.2.3	Y	Y	Y
366	The BA should be able to output the records electronically over a secured connection. The BA should have the ability to transfer the records to a physical storage media that is also removable.	5.3.2.19.2.4	Y	Y	Y

Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)

LEGEND:

ACD	Automatic Call Distributor	kbps	kilobits per second
AEI	AS-SIP End Instrument	LAN	Local Area Network
AMA	Automatic Message Accounting	LDAP	Lightweight Directory Access Protocol
ANAT	Alternative Network Address Types	LDAPv3	Lightweight Directory Access Protocol version 3
ANSI	American National Standards Institue	LSC	Local Session Controller
ASAC	Assured Services Admission Control	MAC	Media Access Control
ASLAN	Assured Services Local Area Network	Mbps	Megabytes per second
AS-SIP	Assured Services Session Initiation Protocol	MFS	Multifunction Switch
ATQA	Attendant Queue Announcement	MFSS	Multifunction Softswitch
B2BUA	Back-to-back User Agent	MG	Media Gateway
BA	Billing Agent	MGC	Media Gateway Controller
BER	Bit Error Rate	MLD	Multicast Listener Discovery
BLV	Busy Line Verification	MLPP	Multilevel Precedence and Preemption
BNF	Backus-Naur Form	Modem	Modulator/Demodulator
C	Conditional	MoIP	Modem over Internet Protocol
C2	Command and Control	MOS	Mean Opinion Score
CAC	Common Access Card	MPLS	Multiprotocol Label Switching
CAS	Channel Associated Signaling	ms	milliseconds
CCA	Call Control Agent	MTU	Maximum Transmission Unit
CCS7	Common Channel Signaling 7	NE	Network Element
CDR	Call Data Record	NM	Network Management
CE	Customer Edge	NMS	Network Management System
CF	Call Forward	OA&M	Operations, Administration, and Maintenance
CH1	Change 1	OCONUS	Outside the Continental United States
CID	Craft Input Device	OSI	Open Systems Interconnect
CND	Calling Number Delivery	OTGR	Operations Technology Generic Requirements
CONUS	Continental United States	PBAS	Precedence Based Assured Services
D-Channel	Data Channel	PBX	Private Branch Exchange
DB	Database	PCD	Precedence Call Diversion
DHCP	Dynamic Host Configuration Protocol	PCM	Pulse Code Modulation
DISA	Defense Information Systems Agency	PEI	Proprietary End Instrument
DISN	Defense Information System Network	PLCP	Physical Layer Convergence Protocol
DN	Directory Number	PRI	Primary Rate Interface
DoD	Department of Defense	PSTN	Public Switch Telephone Network
DS1	Digital Signal Level 1	REL	Release Message
DS3	Digital Signal Level 3	RFC	Request For Communication
DSCP	Differentiated Services Code Point	RTS	Real Time Services
DSN	Defense Switched Network	SAC	Session Admission Control
E2E	End-to-end	SBU	Sensitive, but Unclassified
EBC	Edge Boundary Controller	SCIP	Secure Communications Interoperability Protocol
EC	Echo Canceller	SCS	Session Control and Signaling
EI	End Instrument	SDP	Session Description Protocol
EMS	Element Management System	SG	Signaling Gateway
EO	End Office	SIP	Session Initiation Protocol
ETSI	European Telecommunications Standards Institute	SMEO	Small End Office
FCAPS	Fault, Configuration, Accounting, Performance, and Security	SONET	Synchronous Optical Network
		SRTCP	Secure Real-Time Transport Control Protocol
FoIP	Fax over Internet Protocol	SRTP	Secure Real-Time Transport Protocol
FQDN	Fully Qualified Domain Name	SS	Softswitch
G3 Fax	Group 3 Facsimile	SS7	Signaling System number 7
Hz	Hertz	SUT	System Under Test
IAD	Integrated Access Device	TA	Terminal Adaptor
IAW	In Accordance With	TCA	Threshold Crossing Alert
ICA	Isolated Code Announcement	TCLw	Weighted Terminal Coupling Loss
ID	Identification	TDM	Time Division Multiplexing
ICMPv6	Internet Control Message Protocol for IPv6	TIA	Telecommunications Industry Association
Ie	Equipment Impairment Factor	TLS	Transport Layer Security
IEEE	Institute of Electrical and Electronics Engineers, Inc.	UAC	User Agent Client
IETF	Internet Engineering Task Force	UAS	User Agent Server
IP	Internet Protocol	UC	Unified Capabilities
IPB	IP ASAC Budget	UCR	Unified Capabilities Requirements
IPC	IP ASAC Call Count	UDP	User Datagram Protocol
IPSec	Internet Protocol Security	URI	Uniform Resource Identifier
IPv4	Internet Protocol Version 4	U.S.	United States
IPv6	Internet Protocol Version 6	VBD	Voice Band Data
ISDN	Integrated Services Digital Network	VoIP	Voice over Internet Protocol
ISUP	ISN User Part	VVoIP	Voice and Video over Internet Protocol
ITU-T	International Telecommunications Union – Telecommunication Standardization Sector	WAN	Wide Area Network
		Y	Yes
IWF	Interworking Function		